(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

CORRECTED VERSION

(19) World Intellectual Property Organization
International Bureau

(43) International Publication Date
12 April 2001 (12.04.2001)

PCT

(10) International Publication Number
WO 01/26222 A1

(51) International Patent Classification[7]: H03G 11/00, H04R 29/00, 3/00, H03F 21/00, G10H 1/18, G01R 19/00

(21) International Application Number: PCT/US00/26819

(22) International Filing Date: 29 September 2000 (29.09.2000)

(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:

09/410,338	1 October 1999 (01.10.1999)	US
09/410,616	1 October 1999 (01.10.1999)	US
09/410,612	1 October 1999 (01.10.1999)	US

(71) Applicants and
(72) Inventors: SMITH, Carroll, W. [US/US]; 16709 Amber Lake, Weston, FL 33331 (US). JUREK, Brian, Emerson [US/US]; 3001 South Ocean Drive, #4H, Hollywood, FL 33019 (US). HAZLEWOOD, Harvey, W. [US/US]; 3389 Sheridan, #430, Hollywood, FL 33021 (US).

(74) Agents: SOKOHL, Robert, E. et al.; Sterne, Kessler, Goldstein & Fox P.L.L.C., Suite 600, 1100 New York Avenue, N.W., Washington, DC 20005-3934 (US).

(81) Designated States (*national*): AT, DE, LU, NO, PL, SE.

(84) Designated States (*regional*): European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE).

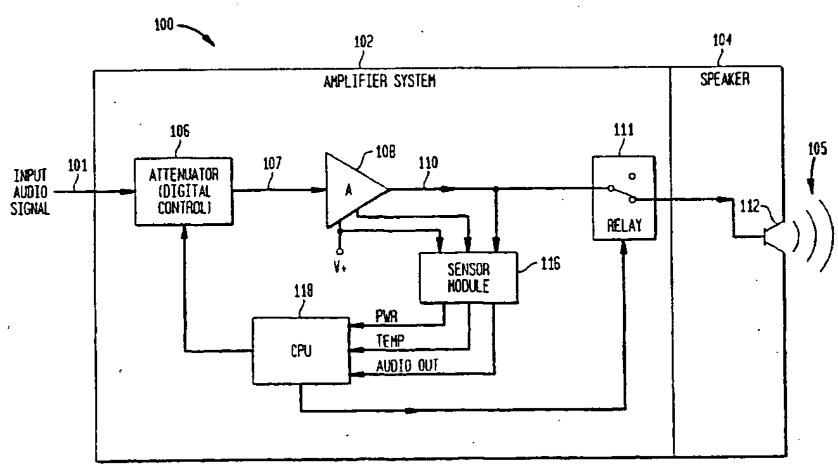
Published:
— *with international search report*

(48) Date of publication of this corrected version: 3 January 2002

(15) Information about Correction:
see PCT Gazette No. 01/2002 of 3 January 2002, Section II

[Continued on next page]

(54) Title: DYNAMIC CLOSED-LOOP AUDIO AMPLIFIER SYSTEM



(57) Abstract: A closed-loop audio amplifier (108) that dynamically amplifiers an input audio signal (101) to drive an audio speaker (104). The amplifier (108) senses the output audio signal (110), amplifier temperature, power supply voltages, and any excessive voltages or currents on the output lines. Based on these sensory inputs, a CPU (118) in the amplifier (108) may dynamically alter the amplifier characteristics to enhance the sound quality produced by the audio speaker and to prevent damage to either the amplifier and/or the audio speaker (104). More specifically, the CPU (118) may tune a digital potentiometer that attenuates the input audio signal (101) prior to amplification to correct compression and temperature concerns. Also, the CPU (118) may disconnect the amplifier (108) from the speaker (104) if the CPU (118) detects an excessive DC voltage or current on the output line, which would indicate a fault with the amplifier (108). The amplifier (108) also includes a DSP for performing signal processing functions including crossover, equalization, time delay, compression, limiting, and gating.

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For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

Dynamic Closed-Loop Audio Amplifier System

Background of the Invention

Field of the Invention

The present invention is generally related to audio amplification. More particularly, the invention relates to a dynamic closed loop audio amplifier system for driving audio speakers.

Related Art

Conventional audio systems include an amplifier that amplifies an audio signal to drive a speaker that is a load connected to the output of the amplifier. The audio system may also include a signal processor that performs any number of signal processing functions on the audio signal prior to driving the speaker. Typically, the amplifier and speaker are connected by some length of speaker wire, which can degrade performance.

Most audio speakers are highly resonant by design. Many high quality speakers are intentionally designed using a vented enclosure with a resonance port to extend the bass response of the speaker. While the extension port does extend the bass response, it also resonates at multiple harmonic frequencies above the desired resonance. The speaker enclosure, either ported or sealed, is also a resonant environment. Odd enclosure dimensions and damping materials can be used to reduce the effects of resonance, but these remedies also impact loudspeaker performance. Furthermore, the speaker drivers (i.e. woofer, tweeter, etc.) are also highly resonant devices. Speaker drivers will vibrate at resonant frequencies and add distortion to the output audio signal unless a high damping factor amplifier is utilized. Often, woofers are designed to extend bass response to the lowest possible frequencies, but this is accompanied by an entire series of

resonances in the higher frequency ranges that add substantial coloration and distortion to the signal. Finally, passive crossover networks also add distortion to the audio signal.

Resonance is directly linked to phase, gain, time, and excursion anomalies. All of these factors lead to non-linearity, high distortion, and the poor reproduction of music or other audible sound. An amplifier with a high damping factor can significantly reduce the resonance problem. These high quality amplifiers typically have a low output impedance that will absorb any back electromotive force (EMF) voltage that is associated with stray or spurious signals, thereby damping these spurious signals. However, wire resistance between the speaker and the amplifier reduces the benefits from of a high damping factor.

Typically, a conventional audio system has an open-loop configuration. In other words, the individual components operate in a stand alone configuration, unaware of what any of the other components are experiencing. As stated above, the primary offender and the highest source of signal distortion is the speaker. Because of the open-loop configuration, distortion produced at the amplifier/speaker interface is not redressed by the electrical components in the audio system. While designers have spent years developing even better speaker systems, there are limits to what is possible within a simplistic open-loop mechanical system. What is needed is a closed-loop amplifier speaker system where the amplifier parameters can be dynamically controlled based on the changing operating conditions.

Summary of the Invention

The present invention is a closed-loop audio amplifier that dynamically amplifies an input audio signal to generate an output audio signal that drives an audio speaker. The amplifier senses the output audio signal, amplifier temperature, power supply voltages, and any DC over-current or voltage on the output line. Based on these sensory inputs, a CPU in the amplifier may dynamically alter the amplifier characteristics to enhance the sound fidelity produced by the audio speaker, and to prevent damage to either the amplifier and/or the audio speaker. In one embodiment, the CPU controls a digital

potentiometer that attenuates the input audio signal prior to amplification to correct power and temperature concerns. Also, the CPU may disconnect the amplifier from the speaker if the CPU detects an excessive DC voltage or current on the output of the amplifier, which would indicate a fault with the amplifier. The result is a closed-loop amplifier and speaker system that can be operated near the maximum output power level, with greater reliability when compared with conventional approaches.

In embodiments of the invention, the amplifier may also include a DSP for performing signal processing functions including crossover, equalization, compression, limiting, gating, and time delay. These DSP functions may be customized for a particular speaker model.

In embodiments of the invention, the amplifier may also include a communications link, which can be used to communicate with an external control device, such as a computer. The invention includes management software that runs on the computer, and can be used to initialize, monitor, and control the amplifier by communicating with the amplifier CPU over the communications link. Furthermore, the computer can be linked to an Internet Web site so that amplifier parameters (including DSP settings that are optimized for a particular speaker model) are downloaded from the Internet Web site to the amplifier CPU.

In embodiments of the invention, multiple amplifier systems are configured to drive an array of associated speakers. For example, in a professional concert application, it is likely that multiple speakers (and therefore multiple amplifiers) will be necessary to generate the required sound level. The management software can be used by an operator to initialize, monitor, and control each individual amplifier through the associated communications link to maximize the overall sound fidelity of the amplifier array.

Embodiments of the invention can be further summarized as follows:

An audio amplifier, said amplifier incorporating a digital CPU wherein said CPU provides supervisory functions overseeing amplifier performance and making corrections to the same as necessary to insure optimal performance.

An audio amplifier wherein a digital CPU monitors the power supply rail voltage, thereby determining the AC line input voltage and configuring the power transformer through a switching means (mechanical or electronic) for proper operation.

An audio amplifier wherein a digital CPU is connected to a digital potentiometer, said digital potentiometer controlling the input signal levels to the amplifier gain stages. The digital CPU is interfaced through a series of A/D converters and buffer devices so as to monitor numerous critical conditions of the amplifier's operation and shall be able to execute corrective measures through the control of the signal levels through the digital potentiometer.

An audio amplifier wherein a digital CPU detects both the voltage available from the amplifier's power supply and the output voltages delivered to the load, said CPU utilizing a digital potentiometer to ride gain on the amplifier signals such that the CPU is able to reduce the gain of the amplifier should the output voltages reach a predetermined percentage of the power supply rail voltages present.

An audio amplifier wherein a digital CPU detects the temperature of the heat sink of the amplifier and compares the heat sink temperature against a software set of values. In the event that the detected temperature shall exceed the first threshold, the CPU shall execute the primary instruction set so as to control the thermal condition. In one instance, the CPU is integrated with a digital potentiometer that controls the gain of the amplifier, said gain will be reduced so as to limit the power output of the amplifier in an attempt to control the heat dissipated within the amplifier. Any such gain change shall be dynamic and time based, the time base being dictated within software. Recovery from the over-temperature state will result in a reversal of the attenuation and return to normal operation so long as the temperature shall remain below the primary threshold. A second software set threshold shall initiate a further CPU response in disconnecting the amplifier from the load, illuminating a "FAILURE" LED indicator and attenuating gain via the digital potentiometer.

An audio amplifier wherein a digital CPU is connected to non-volatile memory and a DSP device, said CPU having a communication means. The CPU handles all communications through a communication means that is integral to the CPU (RX/TX

communications port(s)), and routes DSP specific control instructions to the DSP device in real time or through non-volatile memory.

An audio amplifier wherein a digital CPU monitors output voltage conditions from the amplifier for DC voltages above a software set threshold and time period, said CPU controlling a relay means of disconnecting the amplifier section from the load, thereby protecting the load from damage in the event of an amplifier fault.

An audio amplifier wherein a CPU receives operating information about the conditions of an amplifier, said CPU interfacing to a DSP device through a digital data link, said amplifier having a means of addressing the operating parameters of the DSP device, either in real time or through memory, said amplifier then being able to control numerous DSP functions through a programmed execution. CPU control of DSP functions includes one or more of the following; gain, low pass, high pass and band pass filtering, compression, limiting, gating, equalization and time delay.

An audio amplifier as described herein wherein the digital potentiometer is included within the DSP, said CPU utilizing the digital potentiometer within the DSP for the control functions previously described.

An audio amplifier wherein a digital CPU is able to configure the input and output stages of an amplifier for stereo/mono bridge/parallel bridge modes. The CPU receiving instructions for the configuration of the amplifier either through an interface integral to the amplifier, via non-volatile memory or via a communication means from an external controlling device such as a personal computer.

An audio amplifier wherein a digital CPU is able to control gain of the amplifier through the use of a digital potentiometer, said CPU determining the gain levels from presets within software, an integrated means such as DC potentiometers, or from an external control means communicated via a data link to the CPU.

A dynamically controlled audio amplifier wherein a digital CPU is able to monitor output conditions of the amplifier and adjust gain via a digital potentiometer so as to maximize the linearity of the amplifier under dynamic conditions such as experienced with music. The control means compares the output conditions with software set values or

dynamic amplifier parameters (such as power supply voltages) forming a dynamic closed loop between the output and input of the amplifier.

An audio amplifier fitted with on-board digital memory wherein such memory may be used to store data and/or may be read from or written to from an external means.

An audio amplifier fitted with a DSP, CPU and memory wherein the CPU is able to communicate with an external data source. Said arrangement having the ability to download audio processing algorithms into the resident memory and/or DSP for the purposes of audio enhancement including, but not limited to, psycho-acoustic enhancements.

An audio amplifier wherein a digital means of detecting output levels and frequencies is coupled with a dynamic digital high pass filter section, said filter section being programmed via software for firmware to emulate the characteristics of a loudspeaker driver for the purposes of controlling excursion of a loudspeaker driver. The filter section may have a fixed or variable slope with a variable cut-off frequency such that the level and frequency of energy delivered by the amplifier to the loudspeaker driver is dynamically controlled. Detection of excessive low frequency which would result in excessive driver excursion and possible damage will result in the filter cut-off frequency being shifted higher in frequency so as to negate damaging effects that might otherwise result in driver failure.

An audio amplifier wherein a digital means of detecting output levels and frequencies is coupled with a dynamic digital crossover filter section, said section consisting of a low pass section and high pass section as between a woofer and tweeter, said filter section being programmed via software or firmware to emulate the characteristics of a loudspeaker driver for the purposes of controlling excursion of a loudspeaker driver. Said filter section may have fixed or variable slope with a variable center frequency such that the level and frequency of energy delivered by the amplifier to the loudspeaker drivers is dynamically controlled. In this instance, detection of frequencies and energy levels that would result in high frequency driver damage will initiate an upward shift in the crossover frequency, thereby limiting the excursion of the high frequency driver to safe limits while a corresponding upward shift of the low pass

crossover section allows the low frequency driver to cover the frequency range no longer covered by the high frequency section.

An audio amplifier including a CPU, memory and/or DSP wherein the functionality of the amplifier may be viewed and/or modified remotely using a laptop or PC. Said system shall include a data link between the amplifier and a local PC, said PC being linked via modem or ISP Internet means to a remote PC, said remote PC then having the ability to download historic data or current operational parameters from the amplifier and modify the parameters or software as necessary.

An audio amplifier including a CPU, memory and/or DSP wherein the operating software may be modified, upgraded or replaced via a communications link. The communications link shall allow for a complete upgrade of software within the amplifier without the need to disassemble or modify the hardware in any way. The amplifier includes a read-only memory section with basic boot code and algorithms to verify upload integrity and allow the amplifier to revert to previous software, accept another upload, or enter boot mode in the event of transfer failure.

An audio amplifier with selectable input sensitivities to maximize signal to noise ratio, said sensitivities being controllable by external means such as a PC through a communications link, non-volatile memory settings, or dynamic signal condition.

An audio amplifier utilizing analog means to monitor signal threshold levels which in turn create digital state signals, said digital signals then being routed to an integral CPU. This analog embodiment may use a multitude of comparators whose output(s) may be then routed to the CPU's interrupt pin(s). This arrangement allows for a wide number of operating parameters to be monitored with minimum CPU overhead and only when a corrective condition exists.

Brief Description of the Figures

The present invention will be described with reference to the accompanying drawings, wherein:

FIG. 1 is a diagram that illustrates an amplifier system 100 according to embodiments of the invention;

FIG. 2 is a diagram that illustrates an operational flowchart 200 that is associated with the audio amplifier system 100 in FIG.1;

FIG. 3 is a diagram that illustrates a dual channel audio amplifier 300 according to embodiments of the invention;

FIG. 4 is a diagram that illustrates an operational flowchart 400 that is associated with the audio amplifier 300 in FIG. 3;

FIG. 5 is a diagram that illustrates an operational flowchart 500 that further defines the flowchart 400 in FIG. 4;

FIG. 6A is a functional diagram of the software modules in a CPU 324 according to embodiments of the invention;

FIG. 6B is a diagram that illustrates a DSP 630 according to embodiments of the invention;

FIG. 6C illustrates a diagram of a graphical user interface according to embodiments of the invention according to embodiments of the invention;

FIG. 6D is a diagram that illustrates the functional menus that are available in the graphical user interface;

FIG. 7 illustrates an array of multiple amplifier systems that are controlled using an external control device, according to embodiments of the invention;

FIG. 8 is a diagram that illustrates an operational flowchart 800 that is associated with management software 710 that is shown in FIG.7, according to embodiments of the invention;

FIG. 9A is a diagram that illustrates an operational flowchart 900 that further defines the initialization step 802 in the flowchart 800, according to embodiments of the invention;

FIG. 9B is a diagram that illustrates a log-in menu 930 for the management software, according to embodiments of the invention;

FIG. 9C is a diagram that illustrates a main menu for the management software, according to embodiments of the invention;

FIG. 9D is a diagram that illustrates an amplifier initialization menu for the management software, according to embodiments of the invention;

FIG. 9E is a diagram that illustrates a speaker history file, according to embodiments of the invention;

FIG. 9F is a diagram that illustrates an a job costing menu, according to embodiments of the invention;

FIG. 10 is a diagram that illustrates an operational flowchart 1000 that further defines step 914 in the flowchart 900, according to embodiments of the invention;

FIG. 11A is a diagram that illustrates a signal processing main menu for the management software, according to embodiments of the invention;

FIG. 11B is a diagram that illustrates an equalizer menu for the management software, according to embodiments of the invention;

FIG. 11C is a diagram that illustrates a compressor/limiter/gate menu for the management software, according to embodiments of the invention;

FIG. 11D is a diagram that illustrates a trim and delay adjustment menu for the management software, according to embodiments of the invention;

FIG. 11E is a diagram that illustrates a crossover adjustment menu for the management software, according to embodiments the of the invention;

FIG. 12 is a diagram that illustrates an operational flowchart 1200 for downloading signal processing parameters from an Internet Web site; and

FIG. 13 is a diagram that illustrates an exemplary computer system for operating software aspects of the invention.

Detailed Description of the Preferred Embodiments

1. *Overview*

The present invention is a closed-loop audio amplifier that dynamically amplifies an input audio signal to generate an output audio signal that drives an audio speaker. The amplifier senses the output audio signal, amplifier temperature, power supply voltages, and any DC voltage or current on the output line. Based on these sensory inputs, a CPU in the amplifier may dynamically alter the amplifier characteristics to enhance the sound fidelity produced by the audio speaker, and to prevent damage to either the amplifier and/or the audio speaker. More specifically, the CPU controls a digital potentiometer that attenuates the input audio signal prior to amplification to correct power and temperature concerns. Also, the CPU may disconnect the amplifier from the speaker if the CPU detects an excessive DC voltage or current on the output of the amplifier, which would indicate a fault with the amplifier. The result is a closed-loop amplifier and speaker system that can be operated near the maximum output power level, with greater reliability when compared to conventional approaches.

2. *Dynamic Closed-Loop Audio System*

FIG 1 illustrates an audio system 100 according to one embodiment of the invention. Audio system 100 receives an input audio signal 101 and generates audio sound 105 from speaker 104. Audio system 100 includes amplifier system 102, and speaker 104 having a speaker driver 112. Amplifier system 102 is a closed loop amplifier system that dynamically amplifies the input audio signal 101 to generate an output audio signal 110 that drives the speaker 104. In other words, amplifier system 102 dynamically changes the level of amplification based on the current operating parameters of the amplifier system 102 and the preset values within the CPU 118.

Amplifier system 102 includes the following: attenuator 106, amplifier 108, sensor module 116, relay 111, and CPU 118. Attenuator 106 is digitally controlled by CPU 118, and attenuates the input audio signal prior to amplification by amplifier 108. Sensor module 116 senses operational parameters associated with the amplifier 108, and sends the operational parameters to the CPU 118. More specifically, sensor module 116 senses the following in a preferred embodiment: power supply voltage, temperature, audio output voltage, DC output voltage, and DC output current. Of course, other parameters could be sensed as would be apparent to those skilled in the art(s). Relay 111 connects or disconnects speaker 104 from amplifier 108 under control of CPU 118.

It is advantageous to minimize the distance between amplifier system 102 and speaker 104 in order to minimize the length and resistance of the speaker wire that carries the audio signal 110 from amplifier 108 to the driver 112. Minimizing the length of the speaker wire takes full advantage of the high damping factor associated with the amplifier 108. This is important because a loudspeaker in motion creates an opposing electrical impulse, commonly referred to as back-EMF (electromotive force). The very low output impedance of the amplifier, typically <0.1 ohm, effectively short circuits any back EMF component, thereby damping any stray or spurious signals. In other words, amplifier 108 is better able to sink spurious signals generated by the loudspeaker driver, including signals caused by back EMF and various loudspeaker and enclosure resonances. In one embodiment, the amplifier system 102 is immediately adjacent to speaker 104, as shown in FIG. 1. The amplifier system 102 may be attached to back of the speaker cabinet using fasteners (such as bolts or screws). Alternatively, the amplifier 102 may be fitted in an opening in the speaker cabinet.

The operation of the amplifier system 102 is described with reference to an operational flowchart 200 that is shown in FIG. 2. Flowchart 200 is described as follows.

In step 202, CPU 118 initializes the attenuation for the attenuator 106. In one embodiment, the initial attenuation value may be based on a user input that is loaded into CPU 118. In an alternate embodiment, the initial attenuation value may be "software locked" into the CPU 118 so that the initial attenuation value cannot be changed by unauthorized personnel.

In step 203, attenuator 106 receives the input audio signal 101. In step 204, the attenuator 106 attenuates the input audio signal 101 to generate an audio signal 107, where the amount of attenuation is controlled by CPU 118.

In step 206, amplifier 108 amplifies the audio signal 107 to generate an output audio signal 110 that is coupled to, and drives the speaker 104.

In step 208, sensor module 116 senses operational parameters associated with the amplifier 108, and sends the operational parameters to the CPU 118. More specifically, sensor module 116 senses the following: available power supply voltage for the amplifier 108, temperature of the amplifier 108, voltage of the output signal 110, over current condition for amplifier 108, and any DC output voltage for the amplifier 108.

In step 210, CPU 118 determines if the DC voltage or current at output of amplifier 108 exceeds a threshold. If yes, then there is a fault condition and control flows to step 212. In step 212, CPU 118 disconnects amplifier 108 from speaker 104 by switching relay 111. This prevents damage to amplifier 108 and speaker 104 that can occur when there is a fault mode condition including: high current source or sink from the amplifier output, DC voltage present at the output, initial turn-on/ turn-off transient suppression, and amplifier over-temperature.

In step 214, CPU 118 determines if the amplifier 108 is in a clipped condition or near a clipped condition, based on the output voltage and the available power supply voltage that are detected in step 208. In one embodiment, the voltage of audio output signal 110 is compared to the available power supply voltage. If the output voltage exceeds a percentage threshold of the power supply voltage (e.g. 95%), then clipping are imminent and control flows to step 216. In step 216, the CPU 118 increments the attenuation of the attenuator 106 to reduce the input signal level to the amplifier 108 and bring the amplifier 108 back into linear operation.

In step 218, CPU 118 determines if the amplifier 108 exceeds a temperature threshold. If yes, then control flows to step 220. In step 220, the CPU 118 increments the attenuation of the attenuator 106 to reduce the input signal level to the amplifier 108, and therefore to reduce the temperature of the amplifier 108.

In step 222, the process continues, and control flows back to step 204.

An advantage of the audio system 100 is that it is a closed-loop system. The CPU 118 continuously measures the operational parameters of the amplifier system 102 and dynamically varies the gain based on the operational parameters, which results in improved sound fidelity when compared to conventional audio systems. Conventional audio systems are open-loop and do not vary the amplifier gain based on the amplifier operational parameters.

For example, CPU 118 continuously monitors the output signal voltage of amplifier 108 to determine if the amplifier 108 is in a clipped condition. Amplifier clipping produces noticeable distortion that reduces sound fidelity, and can also cause damage to a speaker driven by the clipped signal. When clipping is detected by the present invention, the CPU 118 increases the input attenuation to bring the amplifier 108 back into linear operation. Conventional audio systems avoid clip induced distortion by under-driving the amplifier so that there is a wide safety margin between the output signal and the power supply rails. In a typical application it is necessary to allow a head room factor of at least 6 dB. This 6 dB allowance relates to one-half power. In other words, twice the equipment is required for the desired sound levels given a 6 dB headroom allowance. This is the result of transients common in musical performances exceeding the power handling capacities of the loudspeaker systems and amplifiers resulting in amplifier clipping and loudspeaker damage. Dynamic control of the clipping conditions through the use of the processing means described herein significantly reduces the headroom allowance required, thereby reducing the amount of equipment required to achieve the desired sound levels while insuring improved reliability.

Furthermore, the audio system 100 has increased reliability over conventional systems because the CPU 118 disconnects the speaker 104 from the amplifier system 102 as soon as a substantial DC output voltage is detected. Also, the CPU 118 continuously monitors the temperature of amplifier 108. The input attenuation is increased when a threshold temperature is reached to reduce the temperature of the amplifier 108 to a normal operating range. Further, the CPU monitors output current to detect a short circuit at the output of the amplifier sections, thereby increasing amplifier reliability. Such an output over-current condition is indicated by a fault LED and can be monitored by a link.

The overall result is that the amplifier system 102 can generate more output signal power than a conventional audio system, with greater reliability than a conventional audio system.

3. *Dual Channel Amplifier System*

FIG. 3 illustrates amplifier system 300 according to embodiments of the invention. Amplifier system 300 is a dual channel version of the amplifier system 102 with some additional functionality. Amplifier system 300 includes: buffer amplifiers 304a,b; pre-amplifiers 358a,b; relays 360a,b; dual digital potentiometer 306; DSP 308; configuration circuits comprising inverting amplifier 310, bridge mono relay 312, and parallel mono relay 314; power amplifiers 316a,b; protection relay 318a,b; CPU 324; sensor module 347 comprising audio preset sensor 326, signal interrupt 330, A/D converter 328, and fault detect 340; communication link 322; memory 331; AC voltage detector 332; user interface 341 having mute switch 334, configuration switch 336, and volume potentiometers 345a, 345b; cooling fan(s) 338; LED(s) display having sleep LED 350, thermal LED 352, protection LED 354, limit LED 356,and vu matrix LED 358; and power supply circuit 342 having voltage select 344 that is fed by AC supply 343, and power supply 346. The operation of amplifier system 300 is further described with reference to flowchart 400, shown in FIG. 4. Flowchart 400 is discussed as follows.

In step 402, CPU 324 receives the initial amplifier settings for the amplifier system 300. These initial settings may include one or more of the following: configuration mode (i.e. stereo mode, full range audio mode, mono bridge mode, or mono parallel mode), initial gain settings, initial DSP settings (i.e. crossover, equalization, time delay, compression, limiting, and gating), and voltage and current threshold settings. In one embodiment, the CPU 324 may receive the amplifier settings from a communication link 322 that is coupled to an external control device, such as a laptop computer. Communication link 322 can be used to communicate information including (initial amplifier setting and current operational parameters) between the CPU 324 and the external control device. Alternatively, the amplifier settings can be received from a user

interface 341 that includes switches mounted on a chasis (not shown) that is associated with the amplifier 300. For example, user interface 341 may include a mute switch 334 to mute the amplifier outputs; a configuration mode switch 336 to select one of stereo mode, mono bridge mode, or mono parallel mode; and volume potentiometers 345a, 345b to adjust the volume of the output signals 320a, 320b, respectively. Volume potentiometers 345a,b are coupled to CPU 324 through A/D converter 328, as shown. In one embodiment, the switches 334, 336, and 345a,b are disabled so that the amplifier settings are solely controlled through the communications interface 322. It is advantageous to disable the switches in a professional concert environment to prevent unauthorized personal from changing the amplifier settings using the switches. Alternatively, the user interface can be a graphical user interface (GUI) that is mounted to the chasis of the amplifier system, such as GUI 650 that is shown in FIGs. 6C and 6D. The details of the GUI 650 will be discussed in a following section.

In step 404, the CPU 324 initializes the amplifier system 300 based on the settings received in step 402. More specifically, CPU 324 sets relays 312 and 314 to configure the amplifier system 300 for one of stereo mode, mono bridge mode, or mono parallel mode. (The configuration modes are discussed further following the discussion of the flowcharts 400 and 500). Additionally, the CPU 324 initializes dual digital potentiometer 306 to set an initial attenuation for the input audio signals, based on the initial gain setting from step 402. Additionally, the CPU 324 initializes the DSP 308 based on the signal processing functions requested in step 402. These signal processing functions may include crossover, equalization, compression, limiting, gating, and time delay, which will be discussed further below. Additionally, the CPU 324 also sets a threshold value for the signal interrupt 330, which monitors the DC output voltage and sends an interrupt signal to the CPU 324 when the threshold is exceeded.

In step 406, buffer amplifiers 304 receive the input audio signals 302a,b at nodes 303a,b, respectively. Buffer amplifiers 304 buffer the rest of the amplifier system 300 from the outside world, as will be understood by those skilled in the relevant art(s). Audio-present sensor 326 senses the outputs of the buffer amplifiers 304a,b using the relays 360a,b to see if an audio signal is present, and notifies the CPU 324. If there is no audio

signal present, then CPU 324 places the amplifier system 300 in a sleep mode by disconnecting the load using relays 318a,b. Muting the input when no audio signal is present saves power and prevents the amplification of unwanted transient input signals and background noise. When the input signals are not being sensed, then the relays 360a,b are configured to couple buffer amplifiers 304a,b to pre-amplifiers 358a,b, respectively, generating audio signals 305a,b.

In step 408, the dual potentiometer 306 attenuates the audio signals 305a,b to generate audio signals 307a,b, where the amount of attenuation is controlled by CPU 324.

In step 410, the DSP 308 preforms digital signal processing functions on the audio signals 307a,b to generate audio signals 309a,b. More specifically, the DSP 308 may preform one or more of crossover functions, equalization functions, time delay functions, compression, limiting, and gating. Crossover (or filtering) may be utilized to separate a full range audio signal received at a single input channel into a high frequency signal and a low frequency signal that are sent out on separate output channels. This avoids damaging a high frequency speaker driver (i.e. tweeter) by driving it with an audio signal that has a high-level bass component. Alternatively, the DSP 308 may receive two input signals at the separate input channels, which are processed separately and then routed to separate output channels. Equalization adjusts the relative signal level between a high frequency signal and the low frequency signal in order to perfect the sound fidelity. Time delay functions are generally utilized in multi-speaker arrays, where the speakers may be separated by some distance. In which case, the audio signal for each speaker may be time delayed by a variable amount, so that the audio signals arrive at their respective speaker in time synchronization with each other. Compression adjusts the signal gain based on output signal power. Limiting assures that the output signal does not exceed some threshold. Gating cuts off input signals that are below a threshold power level, in order to reduce the effect of transient noise signals.

In step 412, power amplifiers 316a,b amplify the audio signals 309a,b to generate audio signals 317a,b, respectively. Audio signals 317a,b may be used to drive one or more speaker drivers, as will be understood by those skilled in the arts.

In step 414, the sensor module 347 detects the current operating parameters for the amplifier system 300. The A/D converter 328 receives signal inputs that represent the current operating parameters including: output voltage and current for power amplifiers 316a,b; available power supply voltage generated by power supply system 342; volume control inputs; and temperature of power amplifiers 316a,b. More specifically, A/D converter 328 directly detects the voltage for the audio signals 317a,b, and the power supply voltages (V_+, V_-) generated by the power supply 346. The current for the audio signals 317a,b is measured by well known current measuring techniques. Volume potentiometers 345a,b generate volume control signals for the two amplifier channels based on user settings, which are then fed to A/D converter 328. Thermometers (not shown), coupled to the power amplifiers 316a,b, measure the current temperature of power amplifiers 316a,b and generate representative signals that are fed to A/D converter 328. A/D converter 328 converts the signal inputs into digital data that can be read by the CPU 324.

Furthermore, the signal interrupt module 330 detects any DC voltage at the output of the power amplifiers 316. The signal interrupt 330 includes a one or more comparators that compare the DC voltage component of the audio signals 317a,b to a threshold value. If the threshold is exceeded, then an interrupt signal is sent to the CPU 324. Similarly, fault detect 340 monitors the current drawn by power amplifiers 316a,b, and sends an interrupt signal to CPU 324 when a current threshold is exceeded.

In step 416, CPU 324 analyzes the operational parameter data received from the A/D converter 328, signal interrupt 330, and fault detect 340, and controls the amplifier system 300 as necessary to optimize performance. Step 416 is further defined by flowchart 500 that is shown in FIG. 5.

Referring now to step 502 in flowchart 500, CPU 324 monitors any interrupt signals received from the signal interrupt 330 and the fault detect 340. If an interrupt is received, then in step 504, the CPU 324 switches the appropriate protection relay 318 to disconnect the corresponding output node 319 from any attached speaker load. This arrangement prevents damage to the loudspeaker in the event of amplifier failure and also prevents damage to the amplifier section in the event of a short circuit at the amplifier

output. In step 506, CPU 324 also turns on a protection LED 354 to indicate that the amplifier system 300 is in the protection mode. In step 508, CPU 324 may also record the protection event in the memory 331, for future reference.

In step 510, CPU 324 determines if audio signals 317a,b are in a clipped condition or are near a clipped condition, which would indicate that the corresponding power amplifier 316 has exceeded it linear operating limits. In one embodiment, this is done by determining if the output voltage swing exceeds a percentage (e.g 95%) of the available power supply voltage that is detected by the A/D converter 328. If yes, then in step 512, the CPU 324 increments the attenuation of the dual digital potentiometer 306 to decrease the input signal level to the power amplifier 316 that is clipped. In step 514, the CPU 324 also turns on a limit LED 356 to indicate that the output signal is limited. In step 516, the CPU 324 may also record the limiting event in the memory 331, for future reference.

In step 518, the CPU 324 determines if the temperature of the power amplifiers 316 are exceeding a threshold temperature level. If yes, then in step 520, the CPU 324 increments the dual digital potentiometer 306 to reduce the input signal level to the power amplifier 316 that is exceeding the threshold temperature level. The CPU may also turn on or adjust the speed of one or more cooling fans 338. In step 522, the CPU 324 turns on thermal LED 352 to indicate the temperature condition. In step 524, the CPU 324 may also record the temperature event in the memory 331 for future reference. When the temperature returns to a normal range, then the CPU 324 may decrease the attenuation of the potentiometer 308, and adjust the cooling fans 338, accordingly.

In step 526, control returns to step 418 in flowchart 400.

In step 418, the CPU 324 uploads the contents of the memory 331 to an external control device (e.g. laptop computer) through the communication link 322. As indicated by steps 508, 516, and 522, the memory 331 contains a running history of amplifier faults recorded during amplifier operation. For example, if a temperature fault was recorded in step 522 during a concert performance, then an operator may upload the operational parameters that were present when the fault occurred. This could be utilized to monitor amplifier performance over time, and to recognize performance trends that could be utilized to avoid similar faults. Step 418 is optional (as indicate by the dotted line

representation), and in one embodiment is done after being polled by the external computer that is connected to the communication link 322. Furthermore, step 420 may be implemented at anytime during flowchart 400, as will be understood by those skilled in the relevant art(s).

In step 422, the process continues, and control flows back to step 406.

3a. Configuration Modes

As discussed in steps 402 and 404, the amplifier system 300 can be operated in multiple configuration modes as controlled by the CPU 324. When operated in stereo (or bi-amp) mode, amplifier system 300 is a dual channel closed loop amplifier that can amplify separate input audio signals 302a and 302b to generate output audio signals 320a and 320b, respectively. [The position of relays 312 and 314, as shown in FIG. 3, are set for stereo mode. Mono bridge mode and mono parallel mode are discussed below.] The input signals 302a and 302b may be for example a low frequency component and a high frequency component of a full range audio signal. In other words, the full range audio signal has already been filtered into a low frequency signal and a high frequency signal that are signals 302a and 302b, respectively. In which case, the output signals 320a,b are coupled to two separate speaker drivers, such as a high frequency driver (i.e. tweeter) and a low frequency driver (i.e. bass). Alternatively, when configured for stereo mode, the amplifier system 300 may receive a single full range audio signal at one of the input nodes 303a or 303b. In which case, the DSP 308 can be programmed to filter the full range audio signal into a low frequency signal and a high frequency signal, which are then separately amplified by amplifier systems 316a and 316b, respectively.

Amplifier system 300 may also be operated in a mono mode, that includes a mono bridge mode and mono parallel mode as controlled by the CPU 324. In the mono bridge mode, amplifier system 300 is configured to receive a single audio signal at node 303a. Relay 312 is switched (from its illustrated position in FIG. 3) so that the input audio signal is fed to both power amplifiers 316a and 316b using an inverting amplifier 310, which inverts the signal going to power amplifier 316b. The output can then be taken from nodes

319a and 319b, doubling the available output voltage that would be obtained if only a single power amplifier 316 was used. In parallel bridge mode, amplifier system 300 also receives a single audio signal at node 303a. Relay 314 is switched (from its illustrated position) so that the audio signal is fed to power amplifiers 316a and 316b, bypassing the inverting amplifier 310. The output can then be taken from nodes 319a and 319b and combined, doubling the available current capability compared to that obtained when using a single power amplifier 316.

3b. *CPU*

As discussed, the CPU 324 supervises or controls the amplifier system 300 based the current operational parameters and any settings received by from communication link 322 or user interface 341. In one embodiment, the CPU 324 is a microprocessor 601 that is illustrated in FIG. 6. An exemplary microprocessor is the Phillips 8052, however other microprocessors could be used as will be understood by those skilled in the arts. Microprocessor 601 includes software modules associated with the operational steps performed by the CPU 324 in flowcharts 400 and 500. More specifically, microprocessor 601 includes a DSP interface 602, potentiometer interface 604, communication interface 606, analysis module 608, sensor interface 610, memory interface 612, user interface receive 614, and LED interface 616. These software modules exist as computer program logic stored in a computer readable medium (including RAM and/or ROM) that is located "on-chip" or "off-chip" relative to the microprocessor 601, as will be understood by those skilled in the arts. Furthermore, the computer program logic may be developed and tested on an external computer, and then downloaded to the microprocessor 601, as will be understood by those skilled in the arts.

A brief description of the software modules in relation to flowcharts 400 and 500 follows. The DSP interface 602 controls the DSP 308 as described in step 404 of the flowchart 400, including controlling the settings for crossover, equalization, time delay, compression, limiting, and gating. The potentiometer interface 604 controls the potentiometer 306 as described in steps 404, 512 and 520. The communication interface

606 sends operational parameter data to the communications link 322 as described in step 418 of flowchart 400. The communications interface also receives operating instructions (including the settings for the amplifier system 300) from the communications link 322 as described in step 404 of the flowchart 400. The sensor interface module 610 receives the current operational parameters from the sensor module 347 as described in step 414. The memory interface module 612 stores/retrieves operational parameters data (including data associated with fault events) to/from the memory 331 as described in steps 508, 516, 524, and 418. User interface module 614 receives any user inputs (for example, volume control) from the user interface 341 as described in step 404. LED interface module 614 controls the LEDs 350-358 as described in step 506, 514, and 522.

3c. DSP

As discussed, the DSP 308 performs signal processing functions including crossover, equalization, compression, limiting, gating, and time delay. FIG. 6B illustrates DSP 630 as an example functional embodiment of the DSP 308. DSP 630 is a dual channel DSP for performing signal processing functions on audio signals 307a,b to generate audio signals 309a,b. Each channel contains an input potentiometer 632, multiple switches 634, a crossover module 635, a 7-band equalizer 636, a compressor/limiter/gate module 638, and multiple delay modules 640.

The various functional components of DSP 630 are described as follows. The input potentiometer 632 and output potentiometer 642 can be used to attenuate the audio signals 307a,b, absorbing the functions of the digital potentiometer 306, if so desired. The 7-band equalizers 636 provide equalization functions between the various frequency ranges of the input audio signals 307. The crossover modules 635 provide filtering functions. The compressor/limiter/gate modules provide for signal compression, limiting, and gating as discussed earlier in step 410. The delay modules 640 provides for signal delay as will be understood by those skilled in the arts. The various switches 634a-m allow for the selection of the various functions of the DSP 630 for each channel. Switch 634m allows for the input signal 307a at node 631a to be switched to the output node at 644b

after being filtered by the crossover module 635a. This allows for a full range signal at node 631a to be filtered into a high and low frequency signal components that appear at separate output nodes 644a and 644b.

The signal processing functionality described herein may implemented using analog circuits, programmable logic circuits, hybrid circuits, application specific circuitry (ASIC), or equivalents thereof, as will be understood by those skilled in the relevant arts based on the discussion herein. These mentioned signal processing implementations are within the scope and spirit of the present invention.

3d. Graphical User Interface

As stated, the present invention may include a graphical user interface that is connected to the CPU, and permits a user to initialize the amplifier system parameters. FIGs. 6C and 6D illustrate an example graphical user interface 650. Graphical user interface 650 includes a graphical display 652, scroll buttons 658, 660, menu button 654, and enter button 656. Menu button 654 allows the user to scroll through the multiple menus that are shown in FIG. 6D. Scroll bars 658 and 660 allow the user to scroll through multiple parameters in a particular menu or to increment values for a particular parameter, as will be understood by those skilled in the relevant arts.

Referring now to FIG. 6D, the GUI 650 includes one or more of the following menus: intro menu 662, security menu 664, DSP menu 666, effects menu 668, mute menu 670, signal processing menu 672, status menu 674, and communication menu 676. After the intro menu 662, the security menu 664 may require the user to enter a password 665 to access the specific parameter menus 666-676. The security menu 664 prevents unauthorized personnel from changing the amplifier system parameters. Alternatively, the security requirement can be disabled to allow more efficient access to the parameter menus. The menus may be provided by the CPU 324 in each amplifier system 300. Alternatively, the menus may be provided by a CPU that is internal to the GUI 650.

Referring now to the specific parameter menus, the DSP menu 666 allows the user to enter or adjust DSP parameters 678, including: crossover, time alignment, equalization,

compression, limiting, phase, gain, and gating. The effects menu 668 allows the user to enter or adjust effects parameters 680 including: 3D surround sound, bass enhancement, THX, Dolby Digital sound, etc. The mute menu 670 allows the user to put the amplifier in mute mode or sleep mode. The signal processing menu 672 allows the user to enter or adjust signal processing parameters including gating, reverberation, echo, and flange/phase parameters. In one embodiment, the DSP menu 666 and signal processing menu 672 are combined into a single DSP related menu, since both are primarily related to functions performed in the DSP. The status menu 674 allows the user to display the status of amplifier system parameters 686 on the graphical display 652, including: output power level, gain, mute status, sleep status, clipping status, compressor status, limiting status, etc. Communication menu 676 allows the user to upload or download data and instructions into the amplifier system. The communication menu 676 also allows the user to transfer instructions from one graphical user interface to another graphical user interface when multiple amplifier systems 300 are configured into an amplifier array. This is advantageous because the amplifier instructions can be entered once at single amplifier, and transferred to multiple other amplifiers without re-entering the instructions at the other amplifiers. The management of an array of amplifiers is further explored in FIG. 7 and the related text. Furthermore, even if there is only a single amplifier, it can be time efficient to enter the desired instructions using the graphical user interface instead of an external computer that uses the communications link 322 to send instructions.

4. ***Amplifier Array Management***

As discussed, amplifier system 300 operates as a closed loop amplifier that dynamically amplifies an input audio signal to generate an output signal that drives an audio speaker. In professional concert applications or loudspeaker applications, it is most likely that an array of multiple speakers, and therefore for multiple amplifiers, will be needed to provide a sufficient sound level. For example, FIG. 7 illustrates an audio system 700 comprising: speakers 702a-n, amplifier systems 300a-n, graphical user interfaces (GUI) 650a-n, an audio source 706, and a computer 708 having management software

710 and a data memory 712. As shown, amplifier systems 300a-n are coupled to corresponding speakers 702a-n that are driven by the audio source 706.

Still referring to Fig. 7, a user manages amplifiers 300a-n using the computer 708. Computer 708 is coupled to the communication link module 322 in each amplifier system 300, to effect communication with the CPU 324 in each amplifier system 300. In one embodiment, computer 708 is a portable computer (such as laptop computer). Computer 708 operates inter-active management software 710 that exists as computer program logic stored in a computer readable medium associated with the computer 708. The management software 710 allows a user to initialize, monitor, and control the settings and operational parameters for amplifier systems 300a-n during a performance. (Herein, the term performance is any application in which audible sound is generated and broadcast, including but is not limited to: professional concert applications, loudspeaker applications, and home or personal audio applications). Management software 710 also allows a user to perform post-production activities including job accounting and post-production reports that are associated one or more performances.

Furthermore, a user may enter settings for the amplifier systems 300a-n using one of the graphical user interfaces (GUI) 650a-n. Each GUI 650 is electrically coupled to the respective CPU 324, and may be physically attached to the chasis of the respective amplifier system 300. Also, all of the GUI 650s are able to communicate with each other. Therefore, the settings entered at one GUI 650 can be transferred to the other GUIs 650, without requiring separate entries at the other GUIs 650.

4a. *Operational Overview*

Management software 710 is further illustrated by flowchart 800 that is depicted in FIG.8. Flowchart 800 gives a broad overview of the management software 710. Additional details are included in the flowcharts and computer "screen shots" that are shown FIGs. 9A-9F, 10, 11A-11E, and FIG. 12. Flowchart 800 is described as follows.

In step 802, a user initializes the amplifier systems 300a-n for a pending performance using the management software 710. More specifically, a user enters the

initial settings for the amplifier systems 300 using the management software 710, and the management software 710 causes the computer 708 to communicate these settings to the CPU(s) 324, for implementation. Step 802 is further described in flowchart 900 in FIG. 9A, which will be discussed in a following section.

In step 804, the audio performance begins, and audio source 706 begins sending audio signals to the amplifier systems 300a-n for broadcast by the speakers 702a-n.

In step 806, the management software 710 periodically polls the CPUs 324 to upload the current operating parameters for their respective amplifier system 300. As discussed earlier, these operating parameters may include temperature, output voltage, output current, and available power supply voltage for the power amplifiers 316 in each amplifier system 300. The CPUs may also upload the current DSP settings, configuration mode settings, etc.

In step 808, the management software 710 displays the status of each amplifier system 300, based on the current operating parameters that were uploaded in step 806. The status for each amplifier system 300 may be displayed in various formats using the management software 710, so that the user can efficiently analyze the current status of each amplifier system 300. In one embodiment, each amplifier system 300 is represented by a computer "icon" whose color varies depending on the status of the amplifier system. For, example, if an amplifier system 300 has violated a temperature threshold or is in a clipped condition, then the associated icon may be displayed as red in color. In contrast, if the status is normal, then the associated icon may be displayed as green. The user may "select" an icon to view the operational parameters that determine the indicated color status for tuning purposes in step 810.

In step 810, the user monitors and adjusts the settings for the amplifier systems 300 using the management software 710, as desired. For example, the user may interactively tune the attenuation or DSP parameters for one or more of the amplifier systems 300 to improve individual amplifier performance, or to improve the overall performance of the amplifier array. For example, if the user sees that a particular amplifier system is approaching a temperature threshold, then the user may intervene and increase the input attenuation to attempt to reduce the temperature of the amplifier system. In

another scenario, the user may decide to tune the DSP parameters for various amplifiers to improve the overall sound fidelity of the amplifier array. In order facilitate inactive use, the management software may include software based "menus" to receive any user inputs for the amplifier settings. Exemplary menus are illustrated in FIGs 9A-9F and 11A-11E, and will be further described in following sections.

In step 812, the management software 710 downloads any new settings to the respective CPUs in the amplifier systems, for implementation.

In step 814, the CPUs 324 implement the new settings for their respective amplifier systems. As depicted in FIG. 8, steps 806-814 may be repeated multiple times during an audio performance, until the audio performance ends.

In step 816, the audio performance ends, and the audio source 706 stops sending audio signals to the amplifier systems 300a-n for broadcast by the speakers 702a-n.

In step 818, the user has the option of whether to do job accounting to analyze the performance. If job accounting is selected, then in step 820, the user performs job accounting functions using the management software 710. After which, the process ends.

An advantage of the present invention is that each amplifier system 300 can be individually controlled and adjusted during a performance based on the current operating parameters, to maximum the sound fidelity of the entire audio system. Conventional audio systems do not have this dynamic closed loop feature, nor do they have ability to remotely control the described individual amplifier parameters within an amplifier array.

4b. Initialization of the Amplifier Array

The initialization of the amplifier systems 300a-n is further described by flowchart 900 that is shown in FIG. 9A. Computer "screen shots" of interactive software menus that are provided by the management software 710 will also be discussed.

Referring now to flowchart 900, in step 901, a user initiates the management software 710. Preferably, the management software 710 is "menu" driven, and has a log-in menu as illustrated by log-in menu 930 that is shown in FIG. 9B. Log-in menu 930 includes a user name window 932, and a user-type window 934. The user-type window

934 is a software "drop box" that allows the user to indicate the level or type-of-use. Exemplary selections in user-type window 934 include user accounting, authorized/trained dealer, untrained dealer, and casual user, etc. Upon entry, the management software 710 may then require the user to enter a password to confirm that the user is allowed to access the level that is indicated in the window 934. Therefore, the overall result is a multi-level security system that permits incremental access to the management software 710, and prevents unauthorized personal from accessing restricted levels of the management software. For example, it may be useful to give accounting personnel access to the job accounting menus in the software 710, without giving them access to the amplifier performance menus. This prevents unauthorized personal from changing the amplifier settings without permission.

Once access is obtained, then the management software 710 displays a main menu 936 as shown in FIG. 9C. The main menu 936 includes multiple “software buttons” for navigating among the sub-menus in the management software. The software buttons include: an initial setup button 938, a protection button 940, a signal processing button 942, a job costing button 943, a customer file button 944, a post production button 946, and a help menu button 948. For initialization, the user selects the initial setup button 938, which leads to the setup menu 950 that is shown in FIG. 9D. Setup menu 950 includes polling button 952, speaker history button 954, current status button 956, and trouble shoot button 958.

In step 902, the user selects the polling button 952, causing the management software 710 to poll the amplifier systems 300a-n. Polling is done to determine which amplifier systems 300 are present and available for operation. The response from each amplifier system 300 includes a fault indication signal that indicates whether a fault has occurred in a recent performance. Faults include temperature events, clipping events, protection events, etc. as described in steps 508, 514, and 522 of the flowchart 500.

In step 904, the management software 710 determines if any faults are indicated for the reporting amplifier systems 300, based on the fault indicator signal, and displays the results. If there are no faults, then control flows to step 914. If there are faults, then control flows to step 906.

In step 906, the user may select the current status button 956, causing the management software 710 to retrieve the current status of the amplifier systems 300 that have faults. More specifically, the computer 708 prompts the CPU 324 to upload the operational parameters (i.e. temperature, output voltage, available voltage supply, and DC output current) that correspond with each reported fault indication, and which are stored in the respective amplifier memory 331.

In step 908, the user may select the speaker history button 954, causing the management software 710 to retrieve historical data for any faulty amplifier systems from the data memory 712. The historical data in memory 712 includes a longer chronology of operational parameter data for the amplifier systems 300 than that retrieved in step 906. In other words, the data retrieved in step 906 may be a described as "snapshot" when compared to the historical data retrieved in step 908. The historical data includes a numerical breakdown of the number and type faults experienced by the amplifier system over its operating life. Furthermore, the historical data may include information about the respective speakers that have been coupled to a current faulty amplifier.

FIG. 9E illustrates an exemplary historical data menu 960 for displaying historical amplifier data and speaker data. Data menu 960 includes the following data fields: amplifier identification; speaker identification; adjustable timer; user defined maximum operating temperature; user defined maximum operating current; fault indicator; number of times the amplifier was powered up; number of times the amplifier was powered down; maximum operating temperature; maximum operating current; number of compressions due to current faults; number of compression due to thermal faults; number of compressions due to output voltage amplitude; number of brown outs; and total clock hours. Other display menus could be used as will be understood by those skilled in the arts based on the discussion given herein.

In step 910, the faults are analyzed for solutions by selecting the trouble shoot button 958. More specifically, the management software 710 compares the historical data from step 908 to the current amplifier status in step 906. Based on the comparison, the user may identify trends in amplifier performance that suggest solutions to the current fault indication. For example, the historical data may indicate that a faulty amplifier system

300 has had multiple temperature (or clipping) events when coupled to the current model (or brand) of speaker 702, but relatively few temperature (or clipping) events when coupled to other types of speakers. This would suggest that the current speaker should be replaced with a different type of speaker. Alternatively, the user can select an automatic trouble-shooting routine to determine a solution for the current amplifier fault.

In step 912, the management software 710 determines if the faults have been resolved. If not then control flows back to step 910 for further trouble-shooting. If yes, then control flows to step 914.

In step 914, the user enters the settings for the amplifier systems 300 using the management software 710. The settings for the amplifier systems 300a-n include gain, maximum temperature, maximum voltage and current, and others. (Signal processing parameters are entered in steps 920 and 922 that are discussed below.) A user enters the amplifier settings through the computer 708 in response to display menus provided by the management software 710, as will be understood by those skilled in the arts. More specifically, the user enters the protection settings by selecting protection button 940, and resets any job cost timers by selecting the job cost button 942. Step 914 is further described in flowchart 1000 that is described as follows.

Referring now to flowchart 1000, in step 1002, the user enters an amplifier label to identify the amplifier system 300 to which the settings are to be applied.

In step 1004, the user has the option of resetting a resettable job cost timer in the CPU(s) 324. The resettable job cost timer records the amount of time that the respective amplifier system 300 is operated during a particular performance. At the end of the performance, the operation time can be uploaded from the CPU(s) 324 and used for job accounting purposes. Each CPU 324 may also contain a permanent job cost timer (that is not resettable), which records the operation time for the amplifier system since being manufactured. In this manner, the permanent and resettable job cost timers are similar to the permanent and resettable odometers that are found in an automobile.

The resettable job cost timer can be reset from the job cost menu 962 that is shown in FIG. 9F. The job cost menu 962 is displayed when the job cost button 943 is selected from the main menu 936. Job cost menu 962 includes reset buttons 964 and 966,

and a record menu 968. The user selects the reset button 962 to reset the job cost timer for a single identified amplifier system 300, and the reset button 964 to reset the job cost timers for all the amplifier systems 300. The record menu 968 includes data fields that record details associated with a specific amplifier rental job, and will be discussed further in a following job-accounting section.

In step 1006, the user enters a maximum temperature threshold for the amplifier system 300. The maximum temperature threshold defines the temperature for the power amplifiers 316 at which corrective action is taken by the CPU 324, as discussed in steps 520-524 of the flowchart 500. The temperature threshold setting (along with voltage and current thresholds that are discussed in step 1008) may be entered by selecting the protection button 940 in the main menu 936.

In step 1008, the user enters the maximum voltage and current thresholds for the amplifier systems 300. The maximum voltage threshold defines the output voltage for the power amplifier 316 at which corrective action is taken by the CPU 324, as discussed in steps 504-508 of the flowchart 500. Likewise, the maximum current threshold defines the output current for the power amplifier 316 at which corrective action is taken by the CPU 324. Furthermore, a threshold may be set for the power supply current that is drawn by the power amplifiers 316, which is used by the fault detector340 to detect excessive current draw.

In step 1010, the user enters a mute time-delay. The mute time-delay defines a time period during which minimal input audio signal is detected by the CPU 324, after which the CPU 324 puts its respective amplifier system 300 into sleep mode. Sleep mode includes disconnecting the outputs using relays 318.

In step 1012, the user enters a mute signal threshold. The mute signal threshold defines a threshold signal level associated with the mute time-delay input. In other words, any audio signal below the mute signal threshold is treated as minimal or non-existent for purposes the mute time-delay.

In step 1014, the user indicates if the settings for all the amplifier systems 300 have been entered. If no, then control flows back to step 1002 to enter the settings for another amplifier system 300. If yes, then control flows back to step 916 in the flowchart 900.

Returning now to flowchart 900, in step 916, the user has the option of entering signal processing parameters by selecting the signal processing button 942 in the main menu 936. If signal processing is not selected then control flows to step 924. If signal processing is selected, then control flows to step 918.

In step 918, the user indicates whether signal processing parameters are to be manually entered through the computer 708. If no, then control flows to step 922, where the signal processing parameters are downloaded from an Internet Web site. If yes, then control flows to step 920, where the user enters the signal processing parameters for the amplifiers 300a-n using menus provided by the management software 710.

FIG. 11A illustrates a exemplary signal processing menu 1102 for receiving signal processing inputs from a user. Signal processing menu 1102 can be used to set the parameters for the dual channel DSP 630 that is illustrated in FIG. 6B. Other DSPs and input formats could be used as will be understood by those skilled in the arts based on the discussion herein. The signal processing menu 1102 includes: indicators 1104 and 1106; input attenuation slider bar 1108; equalization button 1110; compressor/limiter/gate button 1112; trim/delay buttons 1114a,b; post EQ crossover button 1116; master volume slider bar 1118; and amplifier mode box 1120 having a stereo mode selection or a bi-amp mode selection. Indicators 1104 and 1106 indicate whether the respective channels 1 and 2 have active signal inputs. Slider bar 1108 allows the user to set the input attenuation for the associated DSP. Equalization button 1110 allows the user to set the equalization parameters. The compressor/limiter/gate button 1112 allows the user to interactively shape the signal response of the DSP as relating to compression, limiting, and gating functions. The trim/delay buttons 1114 allow the user to set the time delay for each channel in the DSP. Each trim/delay 1114 also includes a gain adjustment for each channel. The post EQ crossover button 1116 allows the user to set the crossover filter parameters for the DSP. The master volume slider bar 1118 allows the user to set the output attenuation for the DSP. Amplifier mode 1120 includes a stereo mode selection and bi-amp selection. Stereo mode is typically used when two output signals are sent to two separate speakers. Whereas, bi-amp mode is typically used when two output signals are sent to two different drivers (e.g. a woofer and tweeter) in the same speaker. In the

bi-amp mode, there may be only one input signal that is filtered into a low frequency signal and a high frequency signal by the DSP, as will be understood by those skilled in the relevant arts. Based on the amplifier mode, the management software 710 indicates the active inputs and outputs, and draws arrows on the menu 1102 to indicate signal direction.

Signal processing menu 1102 is a high level menu that is used to navigate among the lower level signal processing menus. The lower level menus are displayed by selecting the associated software buttons in the menu 1102, as will be discussed in detail below.

FIG. 11B illustrates the equalization menu 1122 that is displayed when the user selects the equalization button 1110. Menu 1122 allows the user to set the equalization parameters for the DSP. Menu 1122 includes multiple filter boxes 1126a-g that allow the user to individually set multiple bandpass filters to adjust the frequency response of the DSP. Each filter box 1126 allows the user to set the filter gain, center frequency, and filter bandwidth (in octaves). The filter gain is set using a slider bar. The center frequency and filter bandwidth are set using software drop boxes with any number of possible selections. The cumulative frequency response of the multiple filters is plotted on the graph 1124, which allows the user to see in real time, the frequency response of any filter adjustments made using the filter boxes 1126. An advantage of being able to adjust multiple bandpass filters is that the user can efficiently compensate for individual speaker resonances that often vary from speaker-to-speaker. Therefore, a particular amplifier does not have to be matched with a particular speaker to maximize overall sound fidelity.

FIG. 11C illustrates a compressor/limiter/gate menu 1130 that adjusts the dynamic range of the DSP. The menu 1130 is displayed when the user selects the compression/limiter/gating button 1112, and allows the user to adjust the compression, limiting, and gating functions for the DSP, all from a single menu. Menu 1130 includes: a compression ratio slider bar 1140 that sets a compression ratio 1142; a compression threshold slider bar 1132 that sets a compression threshold 1134; a peak limiter slider bar 1136 that sets an amplitude peak limit 1138; a gain slider bar 1148 that sets a compression gain 1150; a noise gate slider bar 1152 that sets a noise gate level 1154; and a graph display 1146 that displays a response curve 1144.

The response curve 1144 reflects the relationship between the output signal amplitude (y-axis) vs. input signal amplitude (x-axis) for the DSP. The response curve 1144 allows the user to see in real time, the cumulative effect of any adjustments made using the slider bars in the menu 1130. The x-axis for is labeled from -99dB to 0 dB to reflect the relative input signal level, but the y-axis is unlabeled. (Note that the tick marks along the y-axis are related to the compression ratio 1142, as will be discussed below.) Other display formats could be used as will be understood by those skilled in the arts based on the discussion given herein.

The shape of the response curve 1144 is adjusted by moving the slider bars (in the menu 1130) that were listed above, which changes the associated DSP parameters. More specifically, the input signal is cutoff below the noise gate level 1154, which causes the response curve 1144 to become vertical for input signals below the noise gate level 1154. This allows the user to cutoff any background noise that may be present, at a user defined level. Similarly, the compression threshold 1134 determines the input signal level where the response curve 1144 begins to flatten out. Above the compression threshold 1134, the slope of the response curve 1144 is reduced based on the compression ratio 1142 until the input signal reaches the peak limit 1138. The higher the compression ratio 1142, the lower the slope of the response curve 1144 between interval defined by the threshold 1134 and the peak limit 1138. At the peak limit 1138, the response curve 1144 becomes completely flat reflecting an infinite compression ratio. Finally, the compression gain 1150 moves the entire curve 1144 up and down relative the default settings. (The default response curve is a straight line from the lower left corner to the upper right corner, having a compression ratio of 1 and 0 dB of gain.)

FIG. 11D illustrates a trim and delay adjustment menu 1156 that is displayed when a user selects one of the trim delay buttons 1114. Menu 1156 allows the user individually adjust the attenuation and delay for the two channels of the DSP. The trim and delay adjustment menu 1156 includes the following for each channel of the DSP: an attenuation slider bar 1158, a delay slider bar 1160, first delay box 1162; and a second delay box 1164. The attenuation slider bars 1158 allow the user to individually adjust the attenuation for the channels of the DSP. This is distinct from the attenuation slider bars 1108 and

1118 in the main signal processing menu 1102, as the slider bars 1108 and 1118 affect the attenuation of the two channels by the same amount and do not permit different attenuation settings for each channel. The first delay boxes 1162 display the delay for the respective channels in units of length, and the second delay boxes 1164 display the delay for the respective channels in units of time.

FIG. 11E illustrates the crossover adjustment menu 1166 that is displayed when the user selects the crossover button 1116 on the main menu 1102. The crossover adjustment menu 1166 allows the user to select the center frequency and cutoff frequency for the crossover point in each DSP channel. The menu 1166 also allows the user to select the type of crossover filter to be implemented by the DSP including any one of: bypass, high pass, low pass, and low pass mono.

Returning now to flowchart 900, in step 922, the signal processing parameters may be down-loaded from an Internet Web site if the signal processing parameters are not manually entered through the computer 708. Step 922 is further described in flowchart 1200 that is discussed as follows.

Referring now to flowchart 1200, in step 1202, the management software 710 connects to an Internet web site 714. As will be shown, Internet web site 714 is capable of providing recommended signal processing parameters given a particular speaker brand/model, which can then be down-loaded to the computer 708. Internet web site 714 may also offer optional psycho-acoustic effects that can enhance sound performance, as will be understood by those skilled in the relevant arts.

In step 1204, the user enters a particular speaker brand and/or model number at the web site 714.

In step 1206, the user may also select one or more psycho-acoustic effects such as constant phase, 3D images and effects, servo loop control, inversion effects, and enhanced bass. Psycho-acoustic effects are optional, non-mechanical effects that enhance music enjoyment psychologically.

In step 1208, the web site 714 retrieves signal processing parameters that have been customized to maximize performance for the speaker model entered in step 1204. Preferably, the signal processing parameters have been provided by, or approved by, the

relevant speaker manufacture associated with the model/brand entered in step 1204. Likewise, the web site 714 retrieves the desired psycho-acoustic effects.

In step 1210, the user may cause the web site 714 to download computer readable files containing the signal processing parameters and any psycho-acoustic effects to the computer 708.

In step 1212, the user may cause the management software 710 to disconnect from the Internet web site. After which, control returns to step 924 in flowchart 900.

Returning now to flowchart 900, in step 924, the management software 710 implements settings for the amplifier systems 300a-n. More specially, the management software 710 causes the computer 708 to download the individual settings for each amplifier system 300 to the respective CPU 324. The CPUs 324 then implement the settings for their particular amplifier system 300. An advantage of the present invention is that the settings can be tailored for each amplifier system 300 to improve the overall sound fidelity of the amplifier array. Furthermore, the management software 710 allows the user to enter the settings remotely, instead of at the location of each individual amplifier.

4c. Job Accounting and Post Production Reports

As discussed in step 816 of the flowchart 800, the user has option of performing job accounting functions using the management software 710, including job costing and generating post production reports. In one embodiment, the job accounting functions are directed at tracking data associated with an amplifier renting business, where amplifiers are rented to customers on a temporary basis.

FIG. 9F illustrates an job costing menu 962 (FIG.9F) that is displayed when a user selects the job costing button 942 in the main menu 936 (FIG. 9C). Job costing menu 962 includes a record menu 968 that records details associated with a specific amplifier rental job. As shown, the record menu 968 has the following fields for data entry: amplifier identification, customer ID, job number, job description, job location, estimated job hours, job rental rates, actual job hours, total rental (dollars) earned, and negotiated amount for

job. The user can calculate the expected and actual rental income from renting an amplifier by filling out the appropriate fields. By storing and tracking one or more job records over time, post production reports can be generated by selecting the post production report button 946 in the main menu 936 (FIG. 9C). Specific post production reports that can be generated include: job cost by customer, job; job cost by amplifier, customer; customer by job, amplifier; customer by amplifier, job; amplifier by customer, job; hardware status by amplifier; hardware status by fault type; and hardware status by manufacture.

5. *Exemplary Computer System*

Embodiments of invention may be implemented using hardware, software or a combination thereof and may be implemented in a computer system or other processing system. In fact, in one embodiment, the invention is directed toward a software and/or hardware embodiment in a computer system, such as the computer 708. An example computer system 1302 is shown in FIG. 13. The computer system 1302 includes one or more processors, such as processor 1304. The processor 1304 is connected to a communication bus 1306. The invention can be implemented in various software embodiments that can operate in this example computer system. After reading this description, it will become apparent to a person skilled in the relevant art how to implement the invention using other computer systems and/or computer architectures.

Computer system 1302 also includes a main memory 1308, preferably a random access memory (RAM), and can also include a secondary memory or secondary storage 1310. The secondary memory 1310 can include, for example, a hard disk drive 1312 and a removable storage drive 1314, representing a floppy disk drive, a magnetic tape drive, an optical disk drive, etc. The removable storage drive 1314 reads from and/or writes to a removable storage unit 1316 in a well known manner. Removable storage unit 1316, represents a floppy disk, magnetic tape, optical disk, etc. which is read by

and written to by removable storage drive 1314. As will be appreciated, the removable storage unit 1316 includes a computer usable storage medium having stored therein computer software and/or data.

In alternative embodiments, secondary memory 1310 may include other similar means for allowing computer software and data to be loaded into computer system 1302. Such means can include, for example, a removable storage unit 1320 and an storage interface 1318. Examples of such can include a program cartridge and cartridge interface (such as that found in video game devices), a removable memory chip (such as an EPROM, or PROM) and associated socket, and other removable storage units 1320 and interfaces 1318 which allow software and data to be transferred from the removable storage unit 1320 to the computer system 1302.

Computer system 1302 can also include a communications interface 1322. Communications interface 1322 allows software and data to be transferred between computer system 1302 and external devices 1326. In one embodiment, the external devices 1326 are one or more amplifier systems 300, as shown in FIG. 7. Examples of communications interface 1322 can include a modem, a network interface (such as an Ethernet card), a communications port, a PCMCIA slot and card, etc. Software and data transferred via communications interface 1322 are in the form of signals, which can be electronic, electromagnetic, optical or other signals capable of being received by the communications interface 1322. These signals are provided to the communications interface 1322 via a channel 1324. This channel 1324 can be implemented using wire or cable, fiber optics, a phone line, a cellular phone link, an RF link and other communications channels.

Computer system 1302 may also include well known peripherals 1303 including a display monitor, a keyboard, a printers and facsimile, and a pointing device such a computer mouse, track ball, etc.

In this document, the terms "computer program medium" and "computer usable medium" are used to generally refer to media such as the removable storage devices 1316 and 1318, a hard disk installed in hard disk drive 1312, semiconductor memory

devices including RAM and ROM, and associated signals. These computer program products are means for providing software (including computer programs that embody the invention) and/or data to computer system 1302.

Computer programs (also called computer control logic or computer program logic) are generally stored in main memory 1308 and/or secondary memory 1310 and executed therefrom. Computer programs can also be received via communications interface 1322. Such computer programs, when executed, enable the computer system 1302 to perform the features of the present invention as discussed herein. In particular, the computer programs, when executed, enable the processor 1304 to perform the features of the present invention. Accordingly, such computer programs represent controllers of the computer system 1302.

In an embodiment where the invention is implement using software, the software may be stored in a computer program product and loaded into computer system 1302 using removable storage drive 1314, hard drive 1312 or communications interface 1322. The control logic (software), when executed by the processor 1304, causes the processor 1304 to perform the functions of the invention as described herein.

In another embodiment, the invention is implemented primarily in hardware using, for example, hardware components such as application specific integrated circuits (ASICs), stand alone processors (such as CPU 324), and DSPs. Implementation of the hardware state machine so as to perform the functions described herein will be apparent to persons skilled in the relevant art(s).

In yet another embodiment, the invention is implemented using a combination of both hardware and software.

Conclusion

Example implementations of the systems and components of the invention have been described herein. As noted elsewhere, these example implementations have been described for illustrative purposes only, and are not limiting. Other implementation

embodiments are possible and covered by the invention, such as but not limited to software and software/hardware implementations of the systems and components of the invention. Such implementation embodiments will be apparent to persons skilled in the relevant art(s) based on the teachings contained herein.

While various application embodiments of the present invention have been described above, it should be understood that they have been presented by way of example only, and not limitation. Thus, the breadth and scope of the present invention should not be limited by any of the above-described exemplary embodiments.

What is Claimed Is:

1. A method of dynamically amplifying an audio signal to drive an audio speaker, the method comprising the steps:

(1) receiving the audio signal;

(2) attenuating said audio signal based on a current attenuation value, to generate an attenuated audio signal;

(3) amplifying said attenuated audio signal using an amplifier, to generate an output audio signal capable of driving the audio speaker;

(4) detecting operating parameters associated with said amplifier; and

(5) tuning said current attenuation value in step (2), based on said operating parameters detected in step (4).

2. The method of claim 1, wherein said step (4) comprises at least one of the following:

(a) determining if said amplifier is in a clipped condition;

(b) determining if a temperature of said amplifier exceeds a temperature threshold;

(c) determining if a DC output voltage of said amplifier exceeds a threshold value; and

(d) determining if an output current (AC or DC) of said amplifier exceeds a threshold value.

3. The method claim 2, wherein said step (a) comprises the steps of:

(i) detecting an output voltage swing associated with said amplifier;

(ii) detecting an available power supply voltage associated with said amplifier; and

(iii) determining if said output voltage swing exceeds a threshold percentage of said power supply voltage.

4. The method of claim 2, wherein said step (5) comprises the step of incrementing said current attenuation value if either one of the following conditions is met

(i) said amplifier is in a clipped condition; and

(ii) said amplifier exceeds said threshold temperature value.

5. The method of claim 2, further comprising the step of disconnecting said amplifier from the audio speaker if an excessive DC voltage or current is detected on said output of said amplifier.

6. The method of claim 2, further comprising the step illuminating an appropriate LED if any one of the following conditions is met

(i) said amplifier is in a clipped condition;

(ii) said amplifier exceeds said threshold temperature value; and

(iii) said DC output voltage or current for said amplifier exceeds a threshold value.

7. The method of claim 2, further comprising the step controlling a cooling fan based on said temperature of said amplifier.

8. The method of claim 1, further comprising the step of performing signal processing functions on said attenuated audio signal, said signal processing functions comprising at least one of crossover, equalization, time delay, compression, limiting, and gating.

9. A method of dynamically amplifying an audio signal to drive an audio speaker, the method comprising the steps:

(1) receiving the audio signal;

(2) attenuating said audio signal based on a current attenuation value, to generate an attenuated audio signal;

(3) amplifying said attenuated audio signal using an amplifier, to generate an output audio signal that drives the audio speaker;

(4) detecting operating parameters associated with said amplifier, said detecting step including

(a) determining if said amplifier is in a clipped condition,

(b) determining if said amplifier exceeds a temperature threshold, and

(c) determining if a DC output voltage or current of said amplifier exceeds a threshold value;

(5) tuning said current attenuation value in step (2), based on said operating parameters detected in step (4), said tuning step comprising the step of incrementing said current attenuation value if either one of the following conditions is met

(a) said amplifier is in a clipped condition, and

(b) said amplifier exceeds said threshold temperature value; and

(6) disconnecting said amplifier from said speaker if said DC output voltage or current exceeds a threshold value.

10. A dynamic closed loop audio amplifier system for amplifying an audio signal to drive a speaker, said audio amplifier system comprising:

a digitally controlled attenuator having a current attenuation value, an input of said attenuator receiving the audio signal;

an amplifier, an input of said amplifier coupled to an output of said attenuator, an output of said amplifier coupled to an output node of said amplifier system;

a sensor module, said sensor module detecting operational parameters associated with said amplifier and generating a data signal representative of said operational parameters; and

a CPU, said CPU receiving said data signal and tuning said digitally controlled attenuator based on said operational parameters.

11. The amplifier system of claim 10, wherein said CPU comprises a computer readable medium having computer program logic stored therein, wherein said computer program logic comprises at least one of:

program code means for causing said CPU to determine if said amplifier is in a clipped condition;

program code means for causing said CPU to determine if a temperature of said amplifier exceeds a temperature threshold; and

program code means for causing said CPU to determine if a voltage or current of said amplifier exceeds a threshold.

12. The amplifier system of claim 11, wherein said computer program logic comprises:

program code means for causing said CPU to increment said current attenuation value for said attenuator if either one of the following conditions is met

said amplifier is clipped; and

said amplifier exceeds a temperature threshold.

13. The amplifier system of claim 11, further comprising a relay coupled between said output of said amplifier and said output node of said amplifier system, said relay controlled by said CPU, said computer program logic further comprising program code means for causing said CPU to switch said relay so as to disconnect said output of said amplifier from said output node of said amplifier system when said voltage or current exceeds said threshold.

14. The amplifier system of claim 11, further comprising a cooling fan, said computer program logic further comprising program code means for controlling said cooling fan based on said temperature of said amplifier.

15. The amplifier system of claim 11, further comprising a DSP coupled between an output of said attenuator and an input of said amplifier, said DSP performing signal processing functions on said audio signal, said signal processing functions including at least one of crossover, equalization, time delay, compression, limiting, and gating.

16. The amplifier system of claim 15, wherein said computer program logic of said CPU further comprises program code means for controlling said DSP.

17. The amplifier system of claim 11, further comprising a LED display comprising a plurality of LEDs for displaying a status of said amplifier system, said computer program logic further comprising a means for controlling the illumination of said LEDs based on said operational parameters.

18. The amplifier system of claim 11, further comprising a memory coupled to said CPU, for storing said operational parameters in response to one of a protection event, a compression event, and a temperature event.

19. The amplifier system of claim 10, wherein said sensor module comprises:

an A/D converter for detecting an output voltage of said amplifier and an available power supply voltage, said A/D converter generating said data signal representative of said output voltage and said available power supply voltage.

20. The amplifier system of claim 19, wherein said sensor module further comprises a thermometer coupled to said amplifier and said A/D converter for detecting a temperature of said amplifier, said data signal also representative of said amplifier temperature.

21. The amplifier system of claim 19, further comprising a signal interrupt module coupled to said A/D converter and said CPU, said signal interrupt module generating an interrupt signal that is sent to said CPU when a DC voltage component of said output voltage exceeds a threshold value.

22. The amplifier system of claim 19, further comprising a current measuring means for measuring an output current for said amplifier, said data signal also representative of said output current.

23. The amplifier system of claim 19, further comprising a volume potentiometer coupled to said A/D converter for receiving a volume input for said amplifier, said data signal also representative of said volume input.

24. The amplifier system of claim 19, further comprising a fault detect for detecting when a power supply current associated with said amplifier system exceeds a threshold.

25. The amplifier system of claim 19, further comprising a means for detecting when a load current exceeds a threshold.

26. The amplifier system of claim 10, further comprising one or more configuration relays controlled by said CPU, said configuration relays configuring said amplifier system for one of stereo mode, mono bridge mode, and mono parallel mode.

27. The amplifier system of claim 10, further comprising a communications link coupled said CPU and an external control device, said communications link carrying operational parameters and instructions between said CPU and said external control device.

28. The amplifier system of claim 10, further comprising a signal routing means for routing audio signals.

29. A CPU for controlling an amplifier system, said amplifier system including an amplifier for amplifying an audio signal to drive an audio speaker, said CPU comprising a computer readable medium having computer program logic, said computer program logic comprising:

program code means for causing said CPU to receive a data signal representing operational parameters associated with said amplifier;

program code means for causing said CPU to analyze said operational parameters; and

program code means for causing said CPU to tune said amplifier system based on said analysis of said operational parameters.

30. The CPU of claim 29, wherein said program code means for analyzing said operational parameters comprises at least one of the following:

program code means for causing said CPU to determine if said amplifier is in a clipped condition;

program code means for causing said CPU to determine if a temperature of said amplifier is exceeding a temperature threshold; and

program code means for causing said CPU to determine if a DC output voltage or current is exceeding a threshold value.

31. The CPU of claim 30, wherein said program code means for tuning comprises a program code means for causing said CPU to for increment a current attenuation value if any one of the following conditions are met

said amplifier is clipped, and

said temperature of said amplifier exceeds a threshold value,

wherein said current attenuation value is associated with an attenuator that attenuates said audio signal prior to amplification by said amplifier.

32. The CPU of claim 30, wherein said program code means for tuning comprises a program code means for causing said CPU to switch a relay that disconnects an output of said amplifier from an output of said amplifier system when said DC output current or voltage of said amplifier exceeds a threshold value.

33. The CPU of claim 30, wherein said computer program logic further comprising program code means for tuning comprises a program code for controlling a fan based on said temperature of said amplifier system.

34. The CPU of claim 29, said computer program logic further comprising a program code means for causing said CPU to control a DSP that provides signal processing functions for said amplifier system, said signal processing functions including at least one of crossover, equalization, time delay, compression, limiting, and gating.

35. The CPU of claim 29, said computer program logic further comprising a program code means
for causing said CPU to control the illumination of a LED display, said LED display depicting the status of said amplifier system.

36. The CPU of claim 29, said computer program logic further comprising a program code means for storing and retrieving data to a memory, said data being representative of said operational parameters for said amplifier system.

37. The CPU of claim 29, said computer program logic further comprising a program code means for causing said CPU to control a plurality of configuration relays that configure said amplifier system for one of stereo mode, mono bridge mode, or mono parallel mode.

38. The CPU of claim 29, said computer program logic further comprising a program code means for causing said CPU to control a communications link, said communications link communicating data and instructions between said CPU and an external control device.

39. An audio amplifier, said amplifier comprising a digital CPU having supervisory function for overseeing amplifier performance and making corrections to said amplifier as necessary to insure optimal performance.

40. The audio amplifier of claim 39, wherein said digital CPU monitors a power supply rail voltage to determine an AC line input voltage, and configures a power transformer through a switching means for proper operation.

41. The audio amplifier of claim 39, wherein said digital CPU is connected to a digital potentiometer, said digital potentiometer controlling the input signal levels to gain stages of said amplifier, wherein said digital CPU is coupled to a series of A/D converters and buffer devices so as to monitor numerous critical conditions associated with said amplifier, and wherein said digital CPU executes corrective measures through the control of said input signal levels through the digital potentiometer.

42. The audio amplifier of claim 39, wherein said digital CPU detects an available power supply rail voltage and output voltages that are delivered to a load, said CPU controlling a digital potentiometer to control gain of amplifier signals such that said CPU is able to reduce said gain of said amplifier if said output voltages reach a predetermined percentage of said power supply rail voltage.

43. The audio amplifier of claim 39, wherein said digital CPU detects a temperature of a heat sink of said amplifier and compares said temperature to one or more software thresholds, wherein if said detected temperature exceeds a first threshold then said CPU executes a primary instruction set so as to control said temperature of said amplifier.

44. The audio amplifier of claim 43, wherein said primary instruction set controls an attenuation of a digital potentiometer that controls said gain of said amplifier, said primary instruction set reducing said gain so as to limit a power output of said amplifier and thereby controlling heat dissipated within the amplifier, wherein said gain reduction is dynamic and time based, said time base being dictated within said primary instruction set, wherein said primary instruction set reverses said attenuation when said temperature falls below said first threshold .

45. The audio amplifier of claim 44, wherein said digital CPU executes a secondary instruction set when said amplifier temperature exceeds a second threshold, said secondary instruction set disconnecting said amplifier from a load by switching a relay and illuminating a "FAILURE" LED indicator and attenuating gain using said digital potentiometer.

46. The audio amplifier of claim 39, wherein said digital CPU is connected to a non-volatile memory and a DSP device, said CPU further comprising a communication means, said communications means routing DSP specific control instructions to said DSP device in real time or through said non-volatile memory.

47. The audio amplifier of claim 39, wherein said digital CPU monitors output voltage conditions of said amplifier for DC voltages above a software set threshold and time period, said CPU controlling a relay means for disconnecting said amplifier section from a load, thereby protecting the load from damage in the event of an amplifier fault.

48. The audio amplifier of claim 39, wherein said digital CPU receives operating information about the conditions of said amplifier, said CPU coupled to a DSP device through a digital data link, said CPU having a means of addressing said operating parameters of said DSP device in real time or through memory, said CPU being able to control numerous DSP functions including at least one of the following; gain, low pass filtering, high pass filtering, band pass filtering, compression, limiting, gating, equalization, and time delay.

49. The audio amplifier of claim 48, wherein said DSP comprises a digital potentiometer, said CPU utilizing said digital potentiometer within said DSP to adjust conditions of said amplifier.

50. The audio amplifier of claim 39, wherein said digital CPU configures an input stage and an output stage of said amplifier for one of stereo mode, mono bridge mode, or

parallel bridge mode, wherein said CPU receives instructions for said amplifier configuration from one or more of the following: a user interface that is integral to said amplifier, a memory, and a communication means coupled to an external control device.

51. An audio amplifier of claim 39, wherein said digital CPU is able to control gain of said amplifier using a digital potentiometer, said CPU determining gain levels from one or more of the following: presets within software, an integrated means including DC potentiometers, and an external control means communicated via a data link to said CPU.

52. A dynamically controlled audio amplifier comprising a digital CPU that monitors output conditions of said amplifier and adjusts gain of said amplifier using a digital potentiometer so as to maximize linearity of said amplifier under dynamic conditions such as experienced with music, said CPU having a control means that compares said output conditions with a software set of values forming a dynamic closed loop between an output and an input of said amplifier.

53. The audio amplifier of claim 52, said amplifier further comprising an on-board digital memory, wherein said memory stores data that can be read accessed by an external means.

54. The audio amplifier of claim 52, said amplifier further comprising a DSP and a memory, wherein said CPU communicates with an external data source to download audio processing algorithms into said resident memory or said DSP for audio enhancement including psycho-acoustic enhancements.

55. An audio amplifier comprising a digital means for detecting output levels and frequencies coupled with a dynamic digital high pass filter section, said filter section being programmed using software or firmware to emulate characteristics of a loudspeaker driver for the purposes of controlling excursion of said loudspeaker driver, said filter section may have a fixed or variable slope with a variable cut-off frequency such that a level and

frequency of energy delivered by said amplifier to said loudspeaker driver is dynamically controlled, wherein detection of excessive low frequency energy which would result in excessive driver excursion and possible damage, will result in a filter cut-off frequency being shifted higher in frequency so as to negate damaging effects that might otherwise result in driver failure.

56. An audio amplifier comprising a digital means for detecting output levels and frequencies and a dynamic digital crossover filter section, said crossover section consisting of a low pass section and high pass section as between a woofer and tweeter, said filter section being programmed via software or firmware to emulate characteristics of a loudspeaker driver for the purposes of controlling excursion of said loudspeaker driver, said filter section may have fixed or variable slope with a variable center frequency such that a level and frequency of energy delivered by said amplifier to said loudspeaker driver is dynamically controlled, wherein detection of frequencies and energy levels that would result in high frequency driver damage will cause an upward shift in a crossover frequency, thereby limiting the excursion of said high frequency driver to safe limits while a corresponding upward shift of said low pass crossover section allows a low frequency driver to cover the frequency range no longer covered by said high frequency section.

57. An audio amplifier system comprising an amplifier, a CPU, a memory, and a DSP, wherein the functionality of said amplifier may be viewed and modified remotely using a computer, said system further comprising a communications link between said CPU and said computer, said computer being linked via modem to an ISP Internet means, said computer having the ability to download historic data or current operational parameters from said CPU and modify said parameters or software as necessary.

58. The audio amplifier system of claim 57, wherein operating software for said CPU and DSP may be modified using said communications link without the need to disassemble said amplifier, said memory shall include a read-only memory section with a basic boot code and algorithms to verify upload integrity and allow said amplifier system to revert

to previous software, accept another upload, or a boot mode in the event of transfer failure.

59. The audio amplifier system of claim 57, wherein input sensitivities of said amplifier system are selectable to maximize signal-to-noise ratio, said sensitivities being controllable by said first computer through said communications link, and said settings may stored in said memory.

60. The audio amplifier of claim 57, further comprising an analog monitoring means to monitor signal threshold levels and generate digital state signals, said digital signals then being routed to said CPU, said analog monitoring means including a multitude of comparators having one or more outputs that are routed to one or more corresponding interrupt pins of said CPU, thereby allowing for a wide number of operating parameters to be monitored with minimum CPU overhead.

61. A method of managing a closed-loop audio amplifier system, the method comprising the steps of:

(1) remotely determining a status of said amplifier system; and

(2) remotely controlling said amplifier system based on said status.

62. The method of claim 61, wherein step (1) of remotely determining comprises the step of determining if a fault has occurred, said fault determining step comprising at least one of the following steps:

determining if said amplifier system is in a clipped condition;

determining if a temperature of said amplifier system exceeds a threshold temperature; and

determining if an output voltage or current exceeds a threshold value.

63. The method of claim 61, wherein step (1) of remotely determining comprises the step of polling a CPU in said amplifier system for said status.

64. The method of claim 61, wherein step (1) of remotely determining comprises the step of representing said status of said amplifier system using a computer icon.

65. The method of claim 64, wherein step (1) further comprises the step of changing the color of said computer icon based on said status of said amplifier system.

66. The method of claim 61, wherein step (2) of remotely controlling comprises the steps of:

(a) adjusting settings of the amplifier system based on said status; and

(b) down-loading said adjusted settings to a CPU in said amplifier system.

67. The method of claim 66, wherein step (a) comprises the step of adjusting an attenuation of said amplifier system.

68. The method of claim 66, wherein step (a) comprises at least one of the following:

(i) adjusting a maximum temperature threshold; and

(ii) adjusting a maximum voltage or current threshold.

69. The method of claim 66, wherein step (a) comprises the step of adjusting one or more signal processing parameters of said amplifier system.

70. The method of claim 69, wherein said step of adjusting one or more signal processing parameters comprises at least one of the following:

adjusting equalization parameters for said amplifier system;

adjusting compression parameters for said amplifier system;

adjusting limiting parameters for said amplifier system; and

adjusting time delay parameters for said amplifier system.

71. The method of claim 66, wherein step (a) comprises the step of receiving user inputs to adjust said settings using interactive software menus.

72. The method of claim 61, further comprising the step of:

(3) remotely initializing said amplifier system, said initialization step comprising

(a) polling said amplifier system;

(b) determining if said amplifier system has one or more faults based on said polling step; and

(c) troubleshooting said faults if one or more exist, said troubleshooting step comprising the steps of

(i) retrieving an initial status for said amplifier system;

(ii) retrieving a history for said amplifier system; and

(iii) comparing said current status to said history so as to resolve said one or more faults.

73. The method of claim 61, further comprising the step of:

(3) performing job accounting functions associated with said amplifier system.

74. The method of claim 73, further comprising the step of:

(4) generating post production reports based on said job accounting functions.

75. A method of managing a closed-loop amplifier system comprising the steps of:

(1) receiving a user input to control said amplifier system using an interactive software menu; and

(2) downloading said user input to a CPU in said amplifier system.

76. The method of claim 75, wherein step (1) comprises the step of receiving said user input using a slider bar associated with said software menu.

77. The method of claim 75, wherein step (1) comprises the step of receiving said user input using a software drop box associated with said software menu.

78. The method of claim 75, wherein step (1) comprises the step of receiving a user input to adjust an input attenuation or an output attenuation of said amplifier system.

79. The method of claim 75, wherein step (1) comprises the step of receiving a user input to reset a temporary job cost timer.

80. The method of claim 75, wherein step (1) comprises the step of:

(a) receiving a user input to adjust one or more equalization parameters.

81. The method of claim 80, wherein step (a) comprises the steps of:

(i) receiving a user input to adjust filter parameters for one or more filters using a slider bar or software drop box; and

(ii) displaying a response curve representative of said one or more filters.

82. The method of claim 75, wherein step (1) comprises the step of:

(a) receiving a user input to adjust one or more compression parameters, wherein said compression parameters comprise at least one of compression ratio, compression threshold, and compression gain.

83. The method of claim 82, wherein step (1) further comprises the step of:

(b) displaying a response curve representative of said compression parameters.

84. The method of claim 82, wherein step (1) further comprises the step of:

(b) receiving a user input to adjust one or more limiting parameters and gating parameters.

85. The method of claim 75, wherein step (1) comprises the step of receiving a user input to adjust one or more time delay parameters.

86. The method of claim 75, further comprising the step of:

(3) downloading signal processing parameters from an Internet web site.

87. The method of claim 86, wherein said step (3) of downloading comprises the steps of:

connecting to said Internet web site;

accepting a user input identifying an audio speaker brand or model;

downloading said signal processing parameters from said web site in a computer readable format, wherein said signal processing parameters are associated with said selected speaker.

88. The method of claim 86, further comprising the step of sending said signal processing parameters to said CPU in said amplifier system.

89. A method of managing a closed-loop amplifier system comprising the steps of:

(1) polling a CPU in said amplifier system to determine a status of said amplifier system;

(2) receiving a user input through an interactive software menu to adjust one or more settings for said amplifier system based on said step (1); and

(3) downloading said amplifier settings to said CPU in said amplifier system.

90. A computer program product comprising a computer useable medium having computer program logic stored therein for managing an amplifier system, said computer program logic comprising a program code means for causing a computer to display one or more interactive software menus to receive user inputs to control said amplifier system.

91. The computer program product of claim 90, wherein said software menus include interactive slider bars or software drop boxes to receive said user inputs.

92. The computer program product of claim 90, wherein said computer program logic further comprises a program code means for causing said computer to display a response curve representative of said user inputs.

93. The computer program product of claim 90, wherein said computer program logic further comprises a program code means for causing said computer to transfer said user inputs to a CPU in said amplifier system for implementation.

94. The computer program product of claim 90, wherein said computer program logic further comprises a program code means for downloading signal processing parameters from an Internet web site.

95. The computer program product of claim 94, wherein said program code means for downloading comprises:

program code means for causing said computer to connect to said Internet web site;

program code means for causing said computer to receive a user input identifying an audio speaker brand or model; and

program code means for causing said computer to receive said signal processing parameters from said web site, wherein said signal processing parameters are associated with said identified audio speaker.

96. The computer program product of claim 90, wherein said software menus comprise at least one of the following: a log-in menu, an initial setup menu, a history menu, a job cost menu, a DSP menu, an equalization menu, a dynamic range menu, a trim and delay menu, and a crossover menu.

97. The computer program product of claim 96, wherein said log-in menu includes a user-type input that is associated with security access to said computer program product.

98. The computer program product of claim 96, wherein said DSP menu includes one or more attenuation slider bars.

99. The computer program product of claim 98, wherein said DSP menu further comprises at least one of: an equalization button, a compression/limiter/gate button, a trim delay button, and a post-eq button.

100. The computer program product of claim 96, wherein said equalization menu includes one or more slider bars or drop boxes to receive filter parameters associated with one or more filters.

101. The computer program product of claim 100, wherein said computer program logic further comprises a program code means for generating a response curve representative of said one or more filters.

102. The computer program product of claim 96, wherein said dynamic range menu includes at least one of a compression ratio slider bar, a compression threshold slider bar, a noise gate level slider bar, a peak limit slider bar, and a compression gain slider bar.

103. A computer program product comprising a computer usable medium having computer program logic for managing one or more amplifier systems, said computer program logic comprising:

first program code means for causing a computer to poll an amplifier system to determine a status of said amplifier system;

second program code means for causing said computer to receive user inputs to control said amplifier system based on said status; and

third program code means for causing said computer to download said user inputs to a CPU in said amplifier system.

104. The computer program product of claim 103, said computer program logic further comprising:

fourth program code means for causing said computer to display said status of said amplifier system using a computer icon, wherein a color of said computer icon represents said status.

105. The computer program product of claim 103, said computer program logic further comprising:

fourth program code means for causing said computer to repeat said first, second, and third program code means for multiple amplifier systems.

106. The computer program product of claim 103, said computer program logic further comprising:

fourth program code means for causing said computer to determine if said amplifier system has one or more faults based on said first program code means; and

fifth program code means for troubleshooting said faults if one or more exist, said fifth program code means comprising,

program code means for causing said computer to retrieve a current status for said amplifier system;

program code means for causing said computer to retrieve a history for said amplifier system; and

program code means for causing said computer to compare said current status to said history so as to resolve said one or more faults.

107. An audio system, comprising:

one or more amplifier systems, each amplifier system having a CPU for controlling said respective amplifier system; and

a computer comprising a computer useable medium having computer program logic for managing said one or more amplifier systems, said computer program logic comprising

first program code means for causing said computer to receive user inputs for said one or more amplifier systems, and

second program code means for causing said computer to download said user inputs to said respective CPUs.

108. The system of claim 107, further comprising a communications link coupled between at least one of said CPUs and said computer.

109. The system of claim 107, wherein said first program code comprises a program code means for causing said computer to display software menus to receive said user inputs.

110. The system of claim 107, wherein said computer program logic further comprises a third program code means for causing said computer to poll said one or more amplifier systems for a status of said one or more amplifier systems, including a fault status.

111. The system of claim 110, wherein said computer program logic further comprises a fourth program code means for causing said computer to troubleshoot any faults identified by said third program code means.

112. A graphical user interface (GUI) for controlling a closed-loop audio amplifier system, said graphical user interface comprising:

a display; and

one or more software menus shown on said display to receive user inputs to control settings for said amplifier system.

113. The graphical user interface of claim 112, further comprising:

means for navigating said software menus to insert said user inputs.

114. The graphical user interface of claim 113, wherein said means for navigating comprises at least one of a scroll button, a menu button, and an enter button.

115. The graphical user interface of claim 112, wherein said software menus comprise a security menu, said security menu controlling access to the remaining menus.

116. The graphical user interface of claim 112, wherein said software menus comprise a DSP menu, said DSP menu controlling a DSP in said amplifier system.

117. The graphical user interface of claim 112, wherein said software menus comprise an effects menu, said effects menu allowing a user to enter psycho-acoustic effects for said amplifier system.

118. The graphical user interface of claim 112, wherein said software menus comprise a mute menu, said mute menu allowing a user to put said amplifier system in a mute mode or a sleep mode.

119. The graphical user interface of claim 112, wherein said software menus comprise a status menu, said status menu allowing a user to display a status of said amplifier system on said display, including a fault status.

120. The graphical user interface of claim 112, wherein said software menus comprise a communications menu, said communications menu allowing a user transfer instructions from said GUI to a second GUI.

FIG. 1

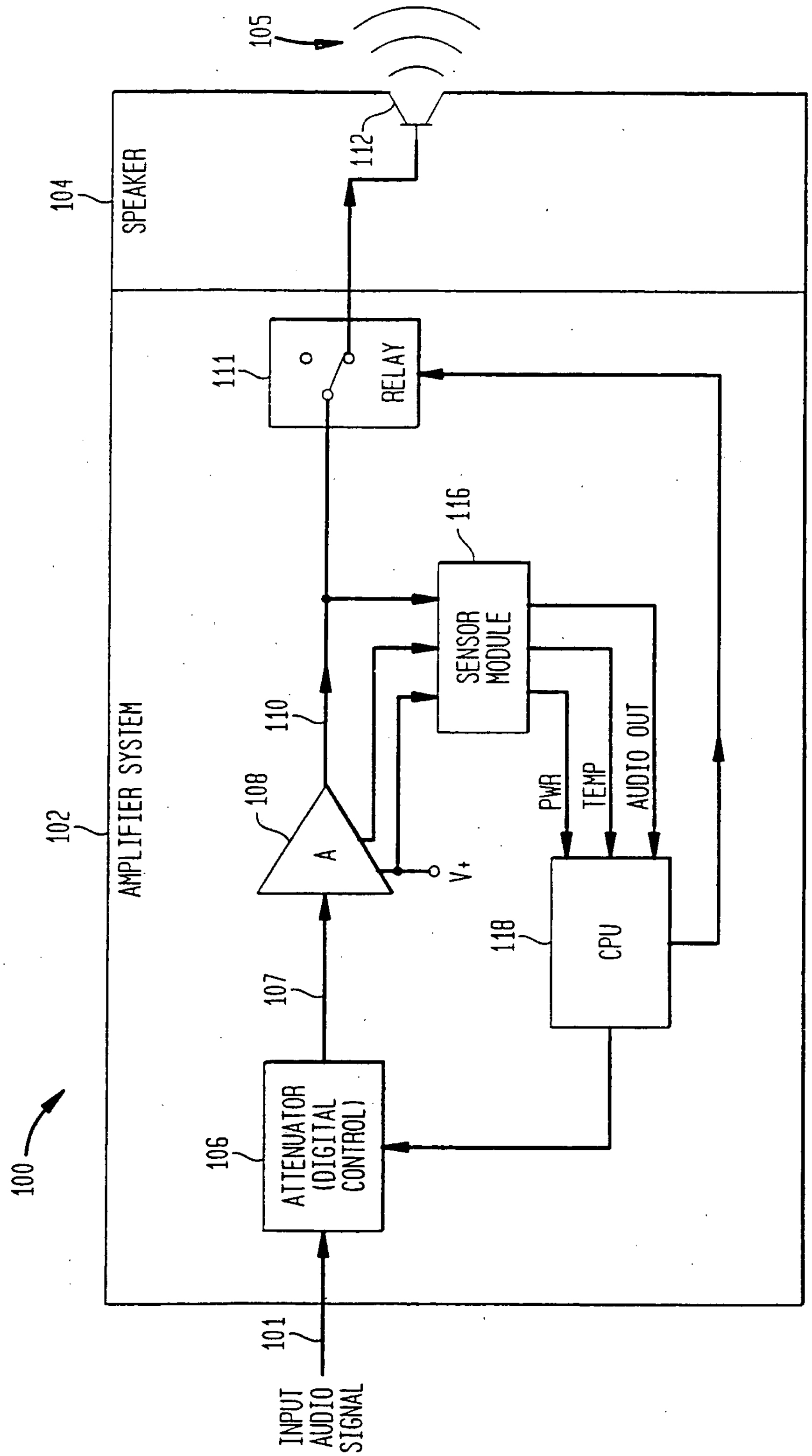
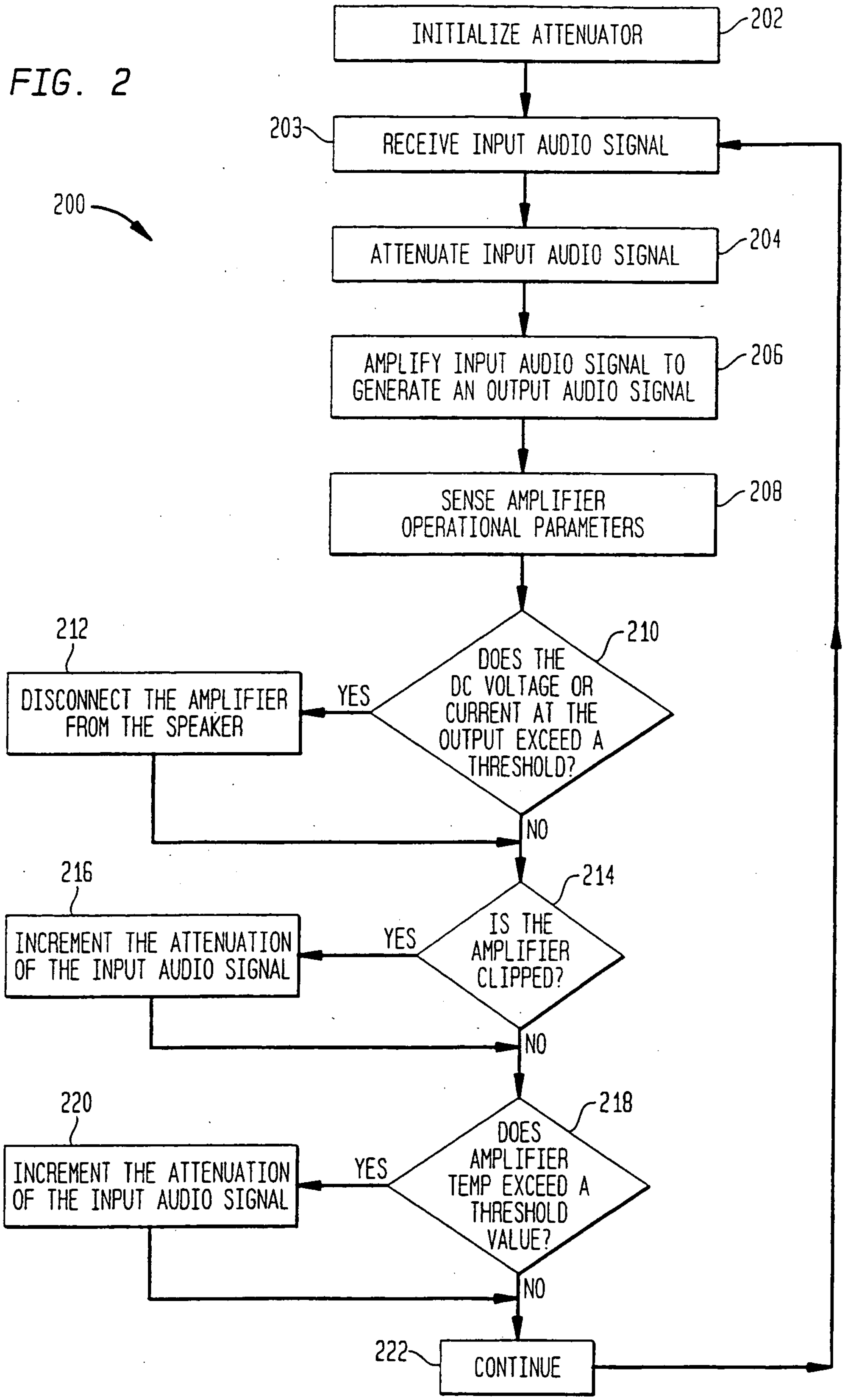

100
105
104
SPEAKER
112
102
AMPLIFIER SYSTEM
111
RELAY
116
SENSOR MODULE
110
108
A
V+
PWR
TEMP
AUDIO OUT
118
CPU
107
106
ATTENUATOR (DIGITAL CONTROL)
101
INPUT AUDIO SIGNAL

FIG. 2



200
202
INITIALIZE ATTENUATOR
203
RECEIVE INPUT AUDIO SIGNAL
204
ATTENUATE INPUT AUDIO SIGNAL
206
AMPLIFY INPUT AUDIO SIGNAL TO GENERATE AN OUTPUT AUDIO SIGNAL
208
SENSE AMPLIFIER OPERATIONAL PARAMETERS
210
DOES THE DC VOLTAGE OR CURRENT AT THE OUTPUT EXCEED A THRESHOLD?
YES
NO
212
DISCONNECT THE AMPLIFIER FROM THE SPEAKER
214
IS THE AMPLIFIER CLIPPED?
YES
NO
216
INCREMENT THE ATTENUATION OF THE INPUT AUDIO SIGNAL
218
DOES AMPLIFIER TEMP EXCEED A THRESHOLD VALUE?
YES
NO
220
INCREMENT THE ATTENUATION OF THE INPUT AUDIO SIGNAL
222
CONTINUE

FIG. 3A

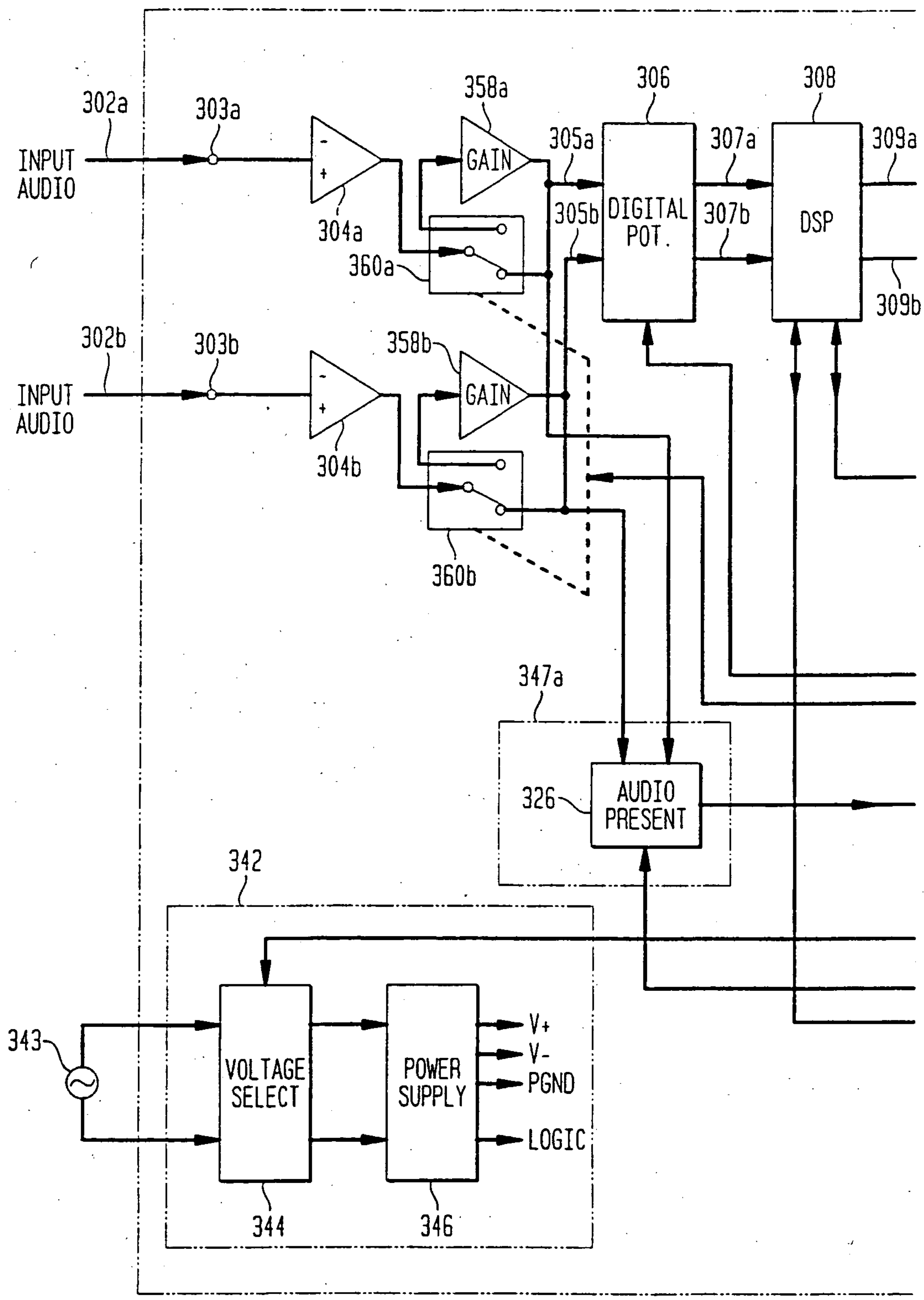

302a
INPUT AUDIO
303a
304a
358a
GAIN
360a
305a
305b
306
DIGITAL POT.
307a
307b
308
DSP
309a
309b
302b
INPUT AUDIO
303b
304b
358b
GAIN
360b
347a
326
AUDIO PRESENT
342
343
VOLTAGE SELECT
344
POWER SUPPLY
346
V+
V-
PGND
LOGIC

FIG. 3B

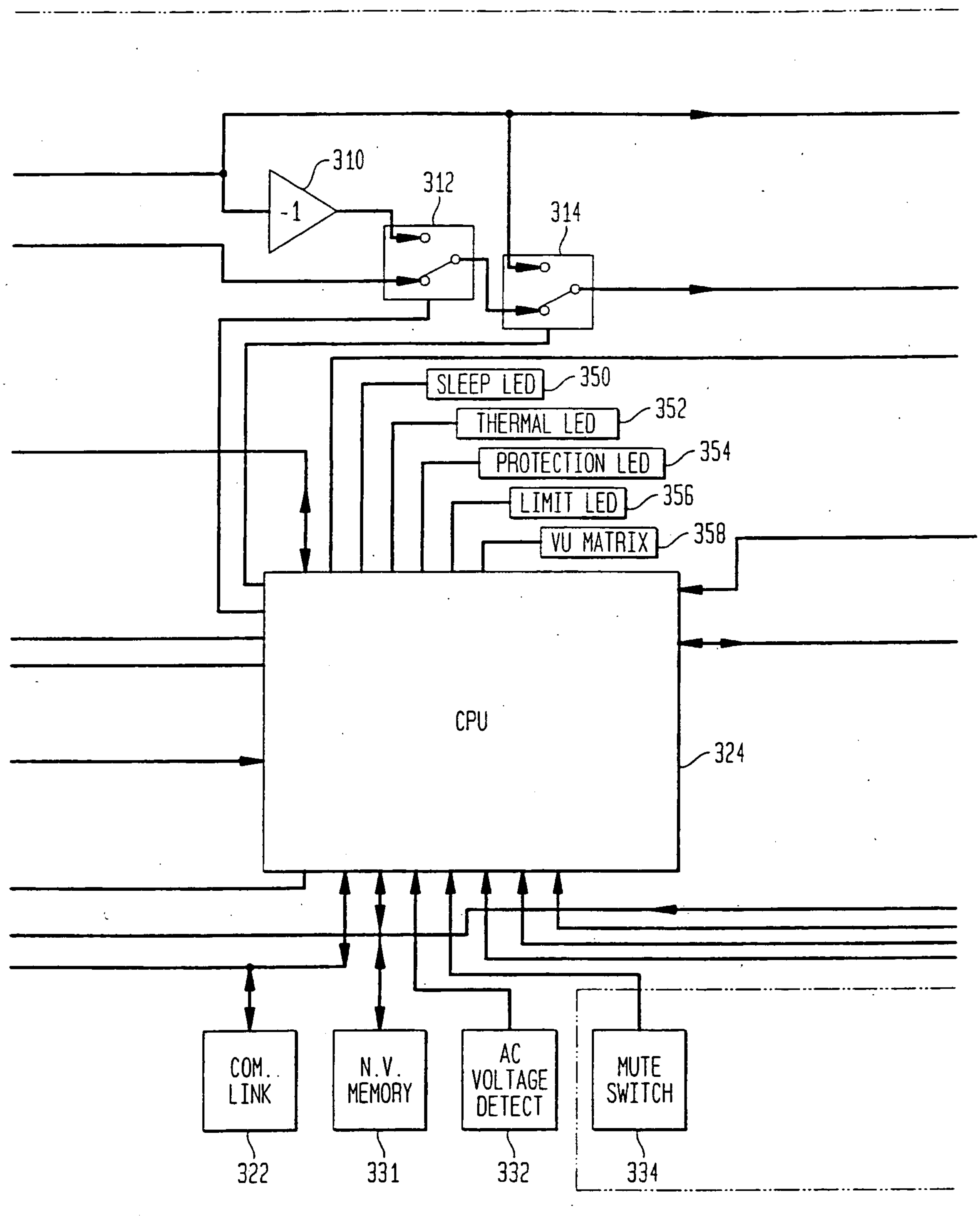

310
-1
312
314
SLEEP LED
350
THERMAL LED
352
PROTECTION LED
354
LIMIT LED
356
VU MATRIX
358
CPU
324
COM. LINK
N.V. MEMORY
AC VOLTAGE DETECT
MUTE SWITCH
322
331
332
334

FIG. 3C

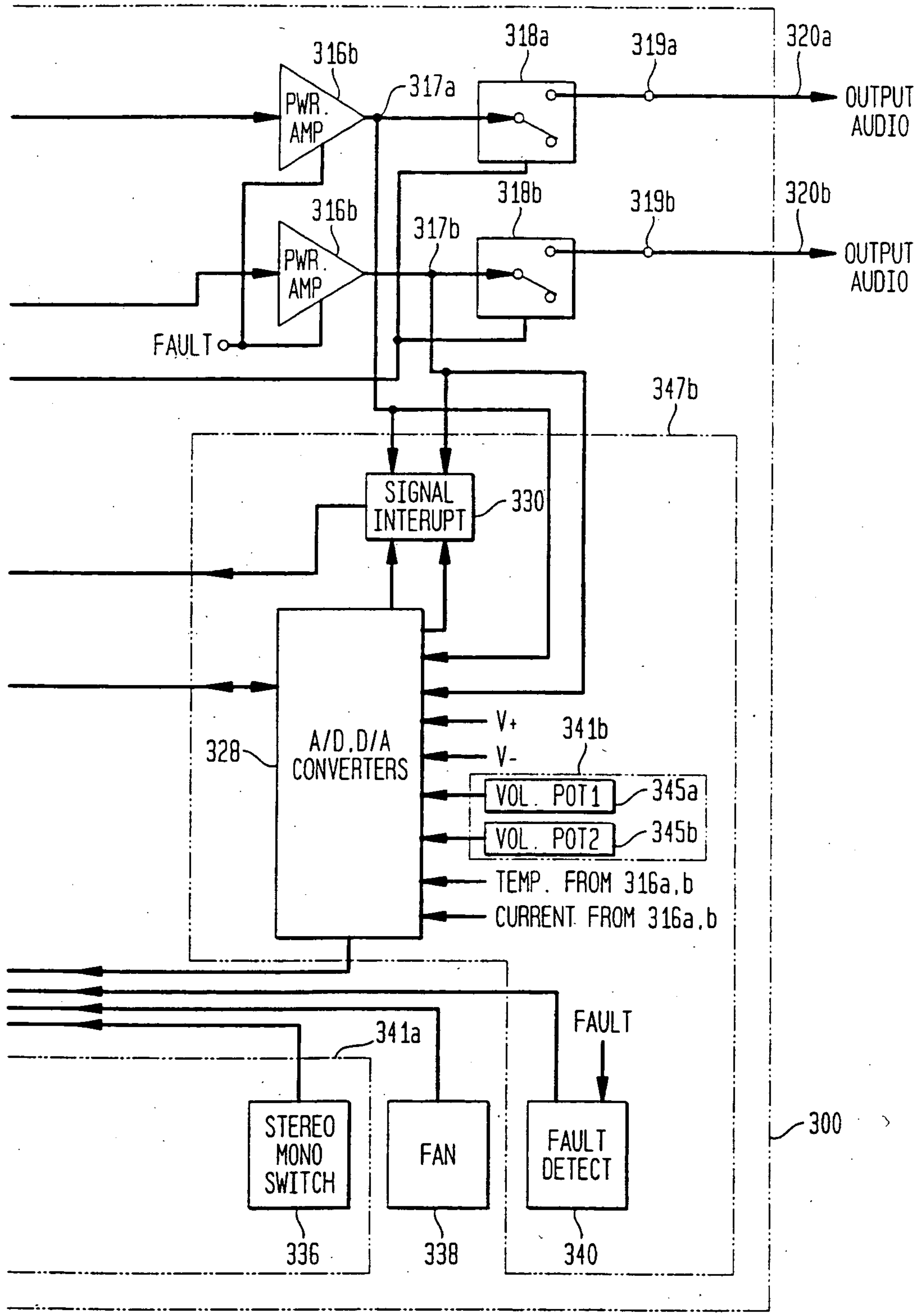

316b
PWR. AMP
317a
318a
319a
320a
OUTPUT AUDIO
316b
PWR. AMP
317b
318b
319b
320b
OUTPUT AUDIO
FAULT
347b
SIGNAL INTERUPT
330
328
A/D.D/A CONVERTERS
V+
V-
341b
VOL. POT1
345a
VOL. POT2
345b
TEMP. FROM 316a,b
CURRENT FROM 316a,b
341a
FAULT
STEREO MONO SWITCH
FAN
FAULT DETECT
300
336
338
340

FIG. 4

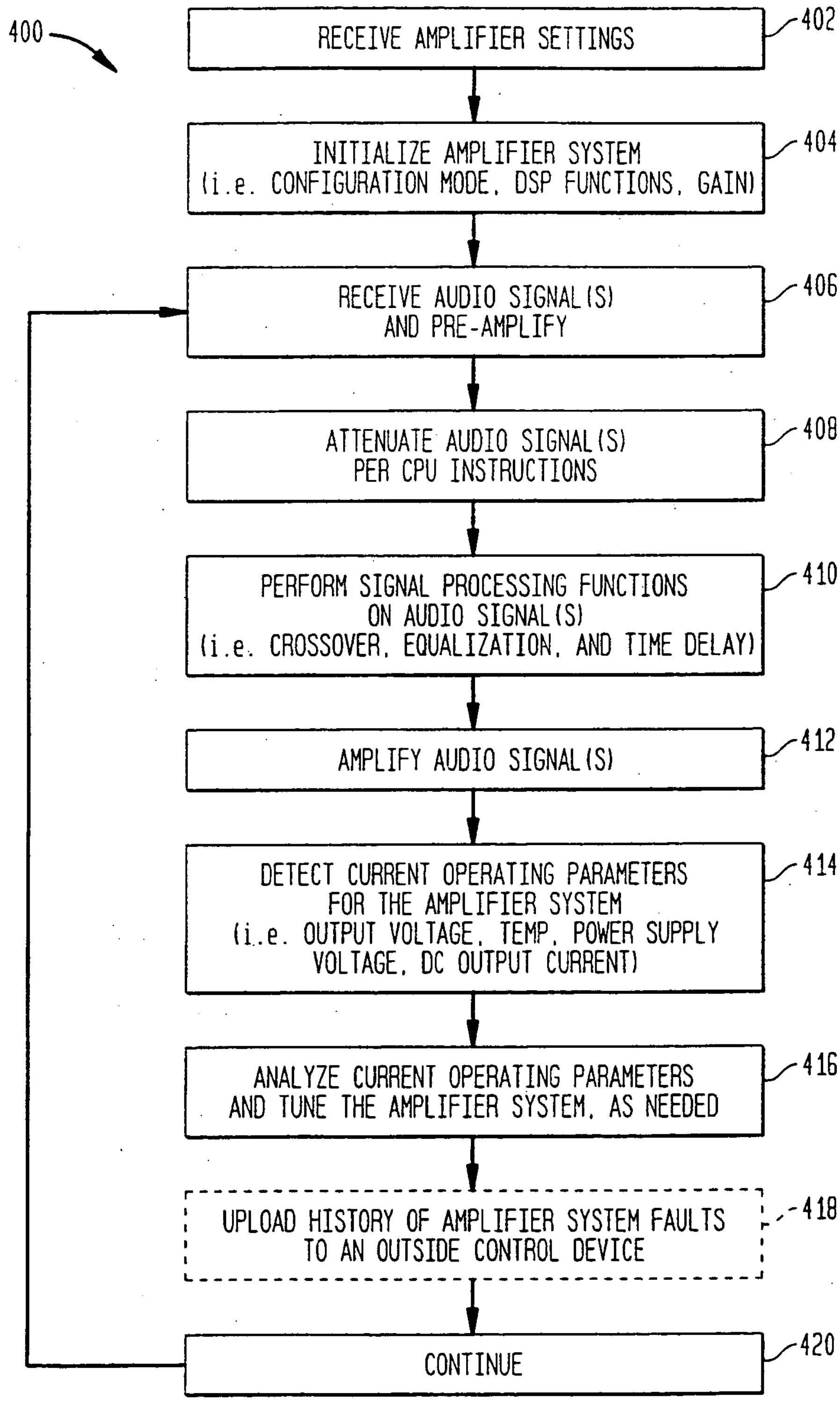

400
RECEIVE AMPLIFIER SETTINGS
402
INITIALIZE AMPLIFIER SYSTEM (i.e. CONFIGURATION MODE, DSP FUNCTIONS, GAIN)
404
RECEIVE AUDIO SIGNAL(S) AND PRE-AMPLIFY
406
ATTENUATE AUDIO SIGNAL(S) PER CPU INSTRUCTIONS
408
PERFORM SIGNAL PROCESSING FUNCTIONS ON AUDIO SIGNAL(S) (i.e. CROSSOVER, EQUALIZATION, AND TIME DELAY)
410
AMPLIFY AUDIO SIGNAL(S)
412
DETECT CURRENT OPERATING PARAMETERS FOR THE AMPLIFIER SYSTEM (i.e. OUTPUT VOLTAGE, TEMP, POWER SUPPLY VOLTAGE, DC OUTPUT CURRENT)
414
ANALYZE CURRENT OPERATING PARAMETERS AND TUNE THE AMPLIFIER SYSTEM, AS NEEDED
416
UPLOAD HISTORY OF AMPLIFIER SYSTEM FAULTS TO AN OUTSIDE CONTROL DEVICE
418
CONTINUE
420

FIG. 5

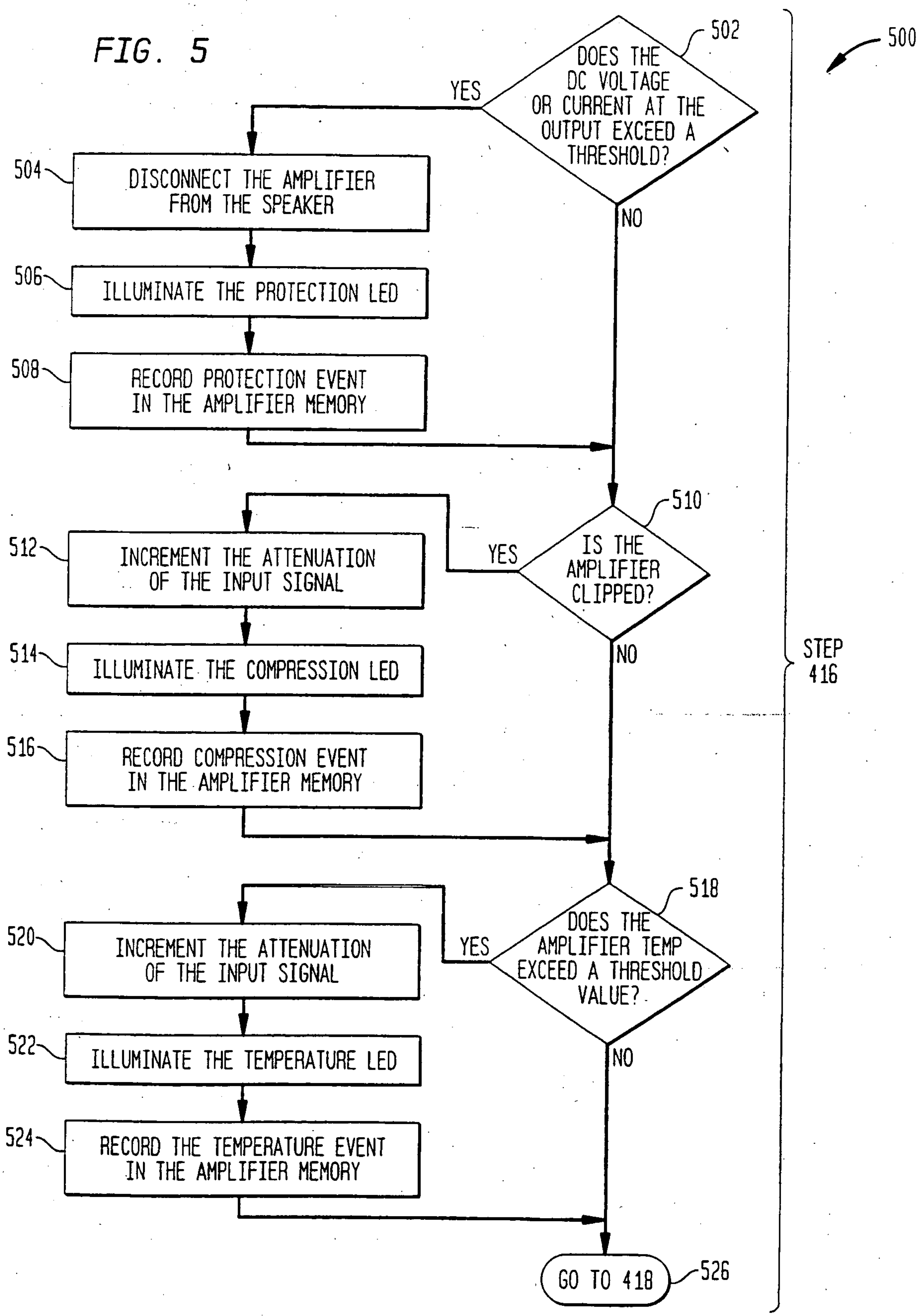
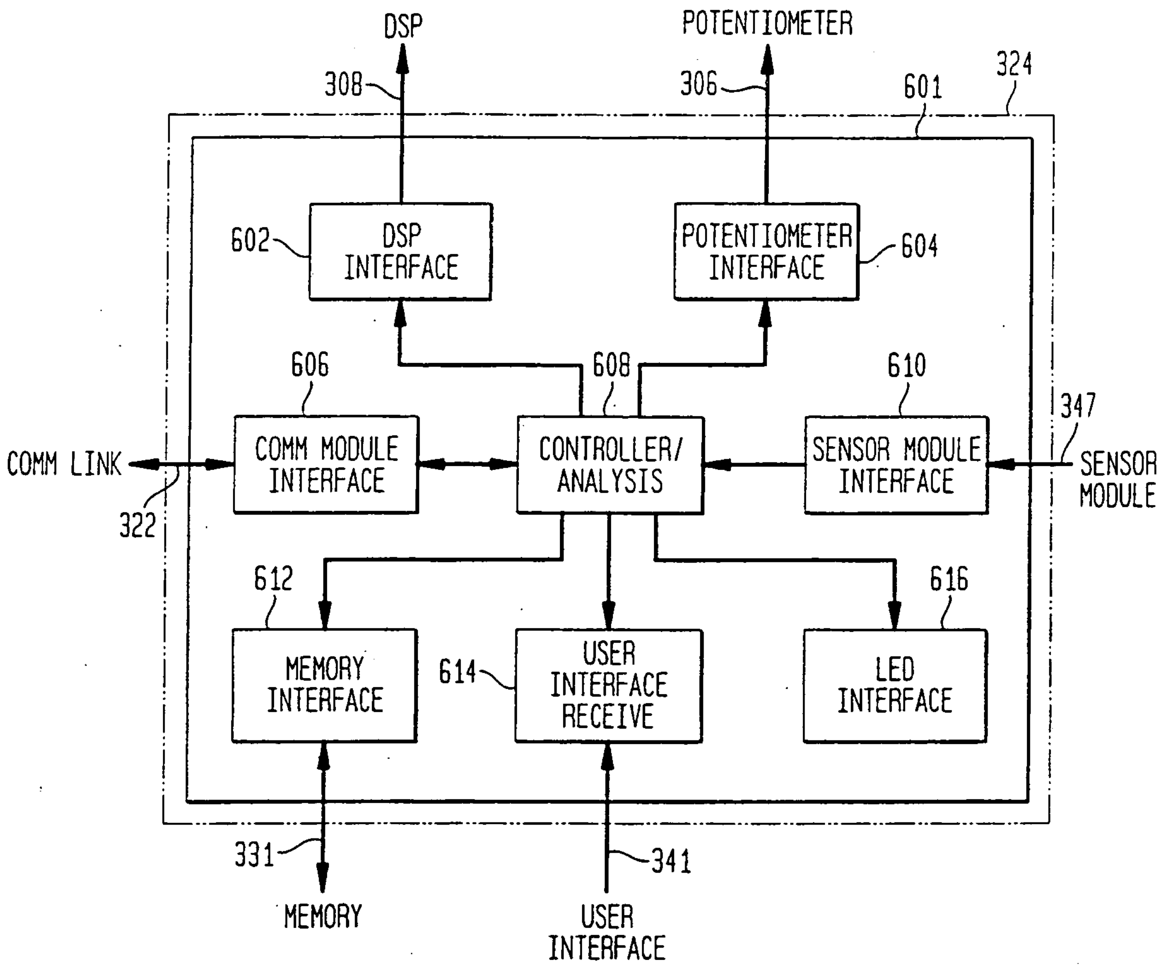

500
502
DOES THE DC VOLTAGE OR CURRENT AT THE OUTPUT EXCEED A THRESHOLD?
YES
NO
504
DISCONNECT THE AMPLIFIER FROM THE SPEAKER
506
ILLUMINATE THE PROTECTION LED
508
RECORD PROTECTION EVENT IN THE AMPLIFIER MEMORY
510
IS THE AMPLIFIER CLIPPED?
YES
NO
512
INCREMENT THE ATTENUATION OF THE INPUT SIGNAL
514
ILLUMINATE THE COMPRESSION LED
516
RECORD COMPRESSION EVENT IN THE AMPLIFIER MEMORY
518
DOES THE AMPLIFIER TEMP EXCEED A THRESHOLD VALUE?
YES
NO
520
INCREMENT THE ATTENUATION OF THE INPUT SIGNAL
522
ILLUMINATE THE TEMPERATURE LED
524
RECORD THE TEMPERATURE EVENT IN THE AMPLIFIER MEMORY
GO TO 418
526
STEP 416

FIG. 6A


DSP
POTENTIOMETER
308
306
601
324
602
DSP INTERFACE
POTENTIOMETER INTERFACE
604
606
608
610
347
COMM LINK
COMM MODULE INTERFACE
CONTROLLER/ ANALYSIS
SENSOR MODULE INTERFACE
SENSOR MODULE
322
612
616
MEMORY INTERFACE
614
USER INTERFACE RECEIVE
LED INTERFACE
331
341
MEMORY
USER INTERFACE

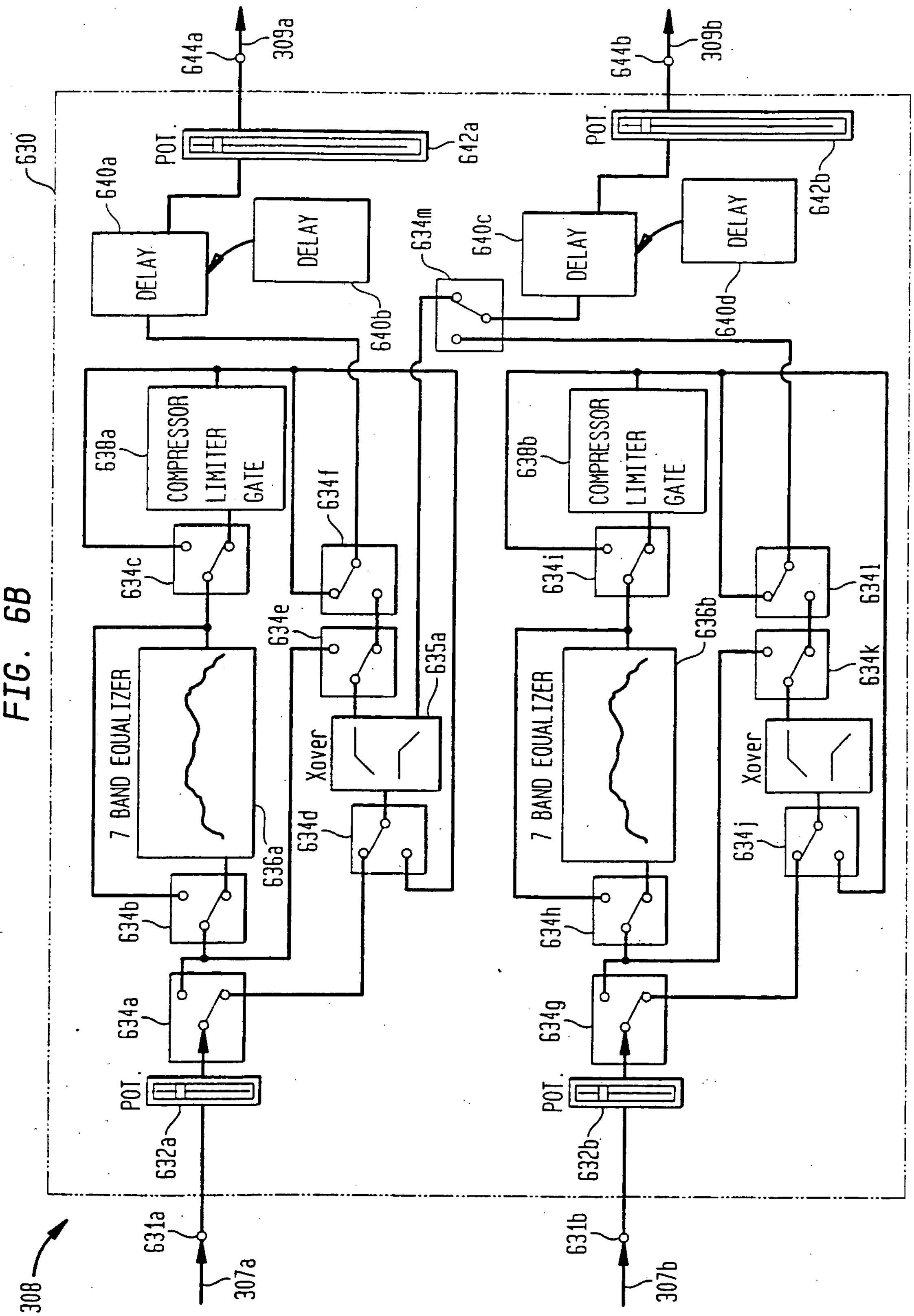

308
630
631a
307a
632a
POT.
634a
634b
7 BAND EQUALIZER
636a
634c
638a
COMPRESSOR
LIMITER
GATE
634d
Xover
634e
635a
634f
DELAY
640a
DELAY
640b
POT.
642a
644a
309a
634m
DELAY
640c
DELAY
640d
POT.
642b
644b
309b
631b
307b
632b
POT.
634g
634h
7 BAND EQUALIZER
636b
634i
638b
COMPRESSOR
LIMITER
GATE
634j
Xover
634k
634l

FIG. 6B

FIG. 6C

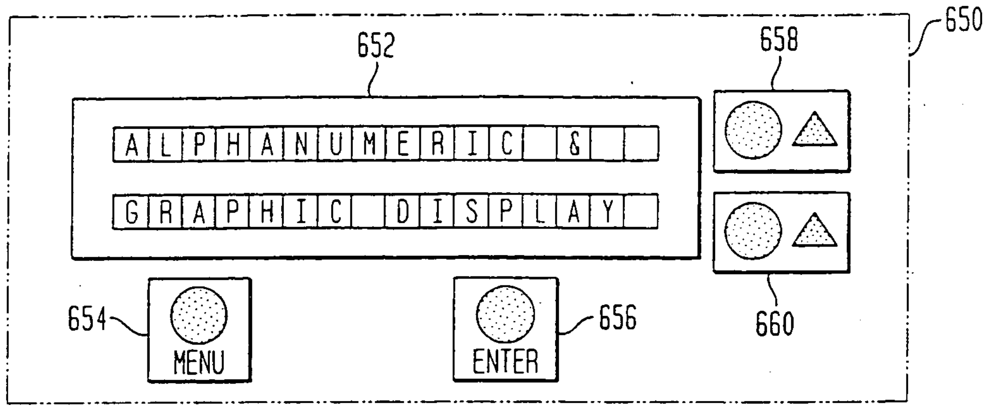

650
652
658
A L P H A N U M E R I C &
G R A P H I C D I S P L A Y
654
MENU
ENTER
656
660

FIG. 6D

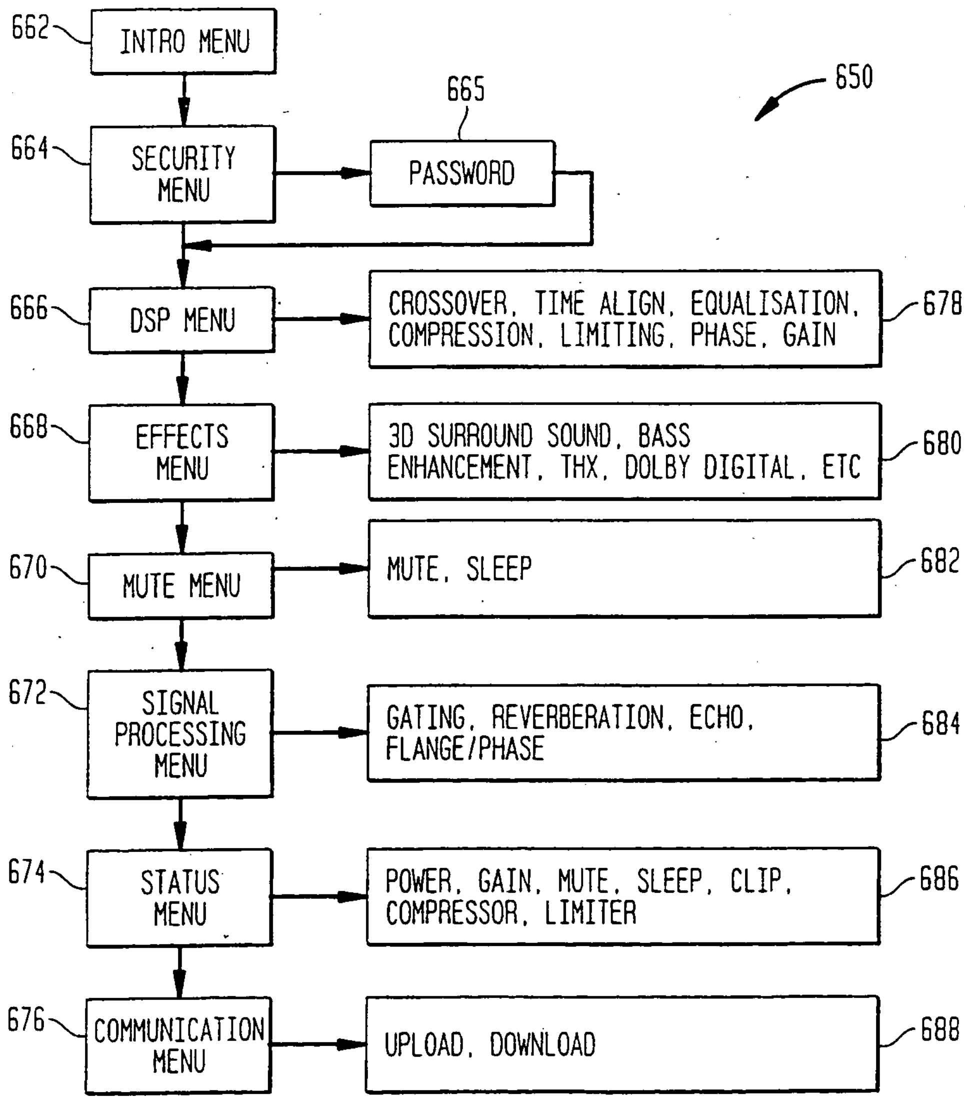

662
INTRO MENU
650
665
664
SECURITY MENU
PASSWORD
666
DSP MENU
CROSSOVER, TIME ALIGN, EQUALISATION, COMPRESSION, LIMITING, PHASE, GAIN
678
668
EFFECTS MENU
3D SURROUND SOUND, BASS ENHANCEMENT, THX, DOLBY DIGITAL, ETC
680
670
MUTE MENU
MUTE, SLEEP
682
672
SIGNAL PROCESSING MENU
GATING, REVERBERATION, ECHO, FLANGE/PHASE
684
674
STATUS MENU
POWER, GAIN, MUTE, SLEEP, CLIP, COMPRESSOR, LIMITER
686
676
COMMUNICATION MENU
UPLOAD, DOWNLOAD
688

FIG. 7

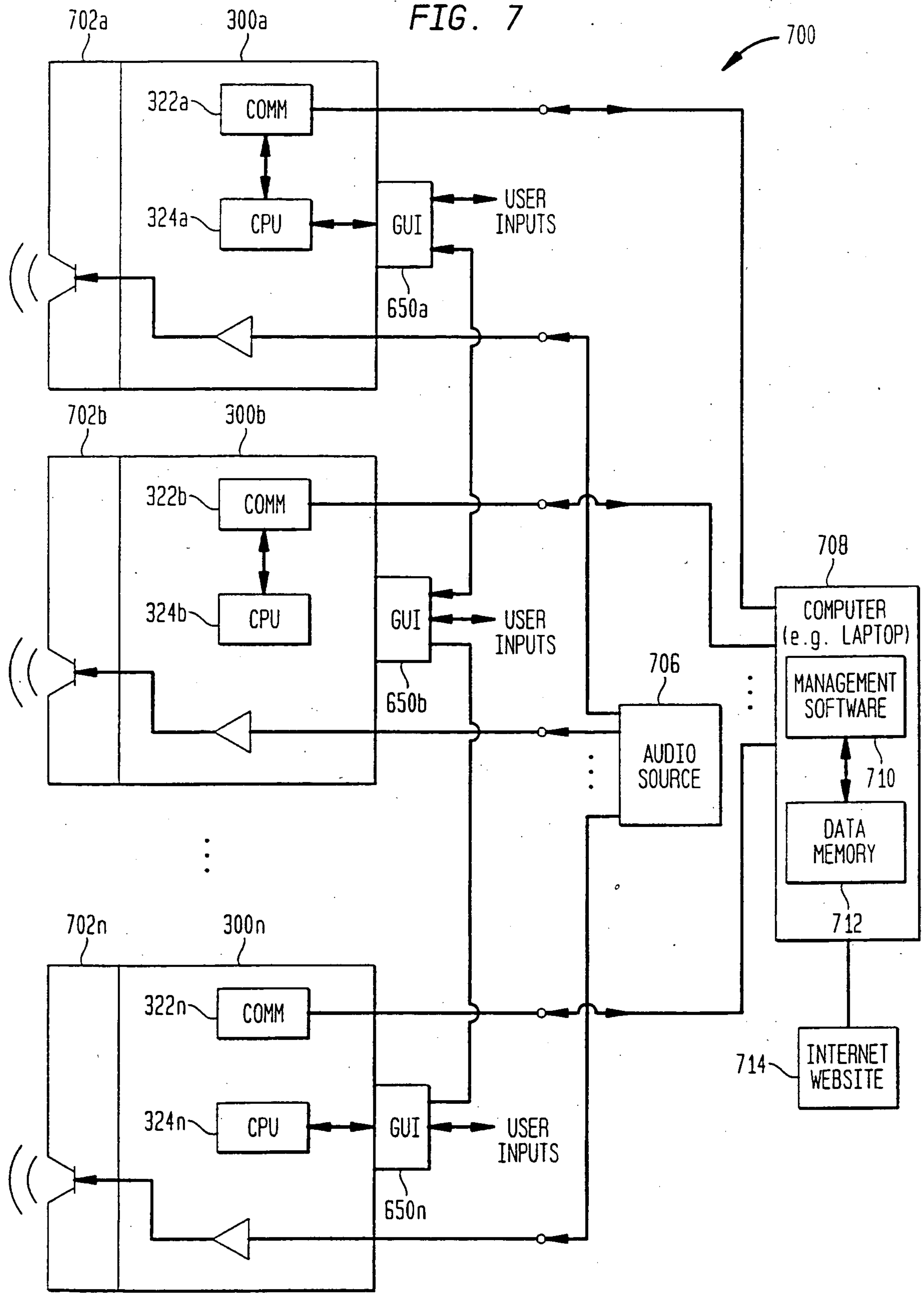

700
702a
300a
322a
COMM
324a
CPU
GUI
USER INPUTS
650a
702b
300b
322b
COMM
324b
CPU
GUI
USER INPUTS
650b
706
AUDIO SOURCE
708
COMPUTER (e.g. LAPTOP)
MANAGEMENT SOFTWARE
710
DATA MEMORY
712
702n
300n
322n
COMM
324n
CPU
GUI
USER INPUTS
650n
714
INTERNET WEBSITE

FIG. 8

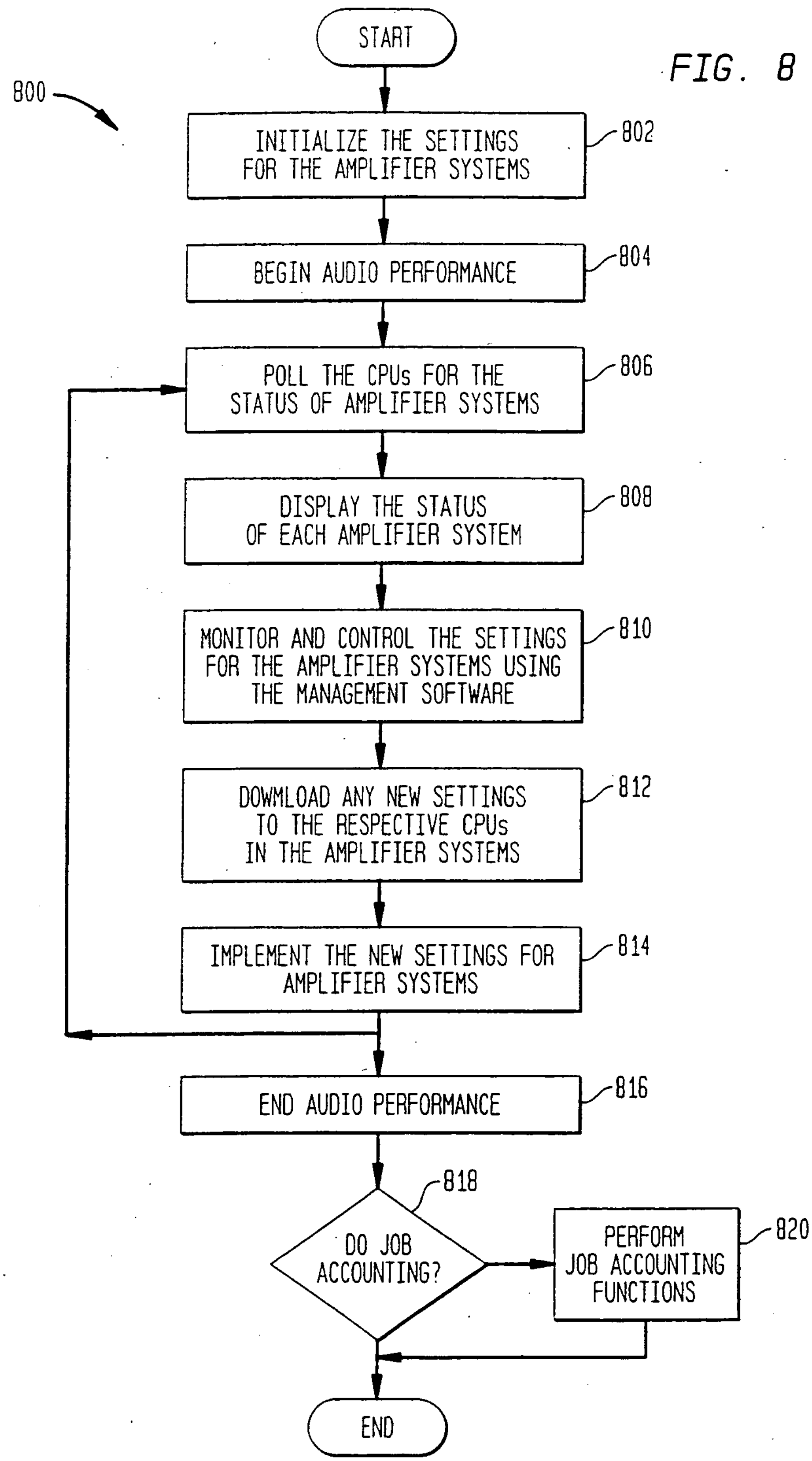

START
800
INITIALIZE THE SETTINGS FOR THE AMPLIFIER SYSTEMS
802
BEGIN AUDIO PERFORMANCE
804
POLL THE CPUs FOR THE STATUS OF AMPLIFIER SYSTEMS
806
DISPLAY THE STATUS OF EACH AMPLIFIER SYSTEM
808
MONITOR AND CONTROL THE SETTINGS FOR THE AMPLIFIER SYSTEMS USING THE MANAGEMENT SOFTWARE
810
DOWMLOAD ANY NEW SETTINGS TO THE RESPECTIVE CPUs IN THE AMPLIFIER SYSTEMS
812
IMPLEMENT THE NEW SETTINGS FOR AMPLIFIER SYSTEMS
814
END AUDIO PERFORMANCE
816
DO JOB ACCOUNTING?
818
PERFORM JOB ACCOUNTING FUNCTIONS
820
END

FIG. 9A

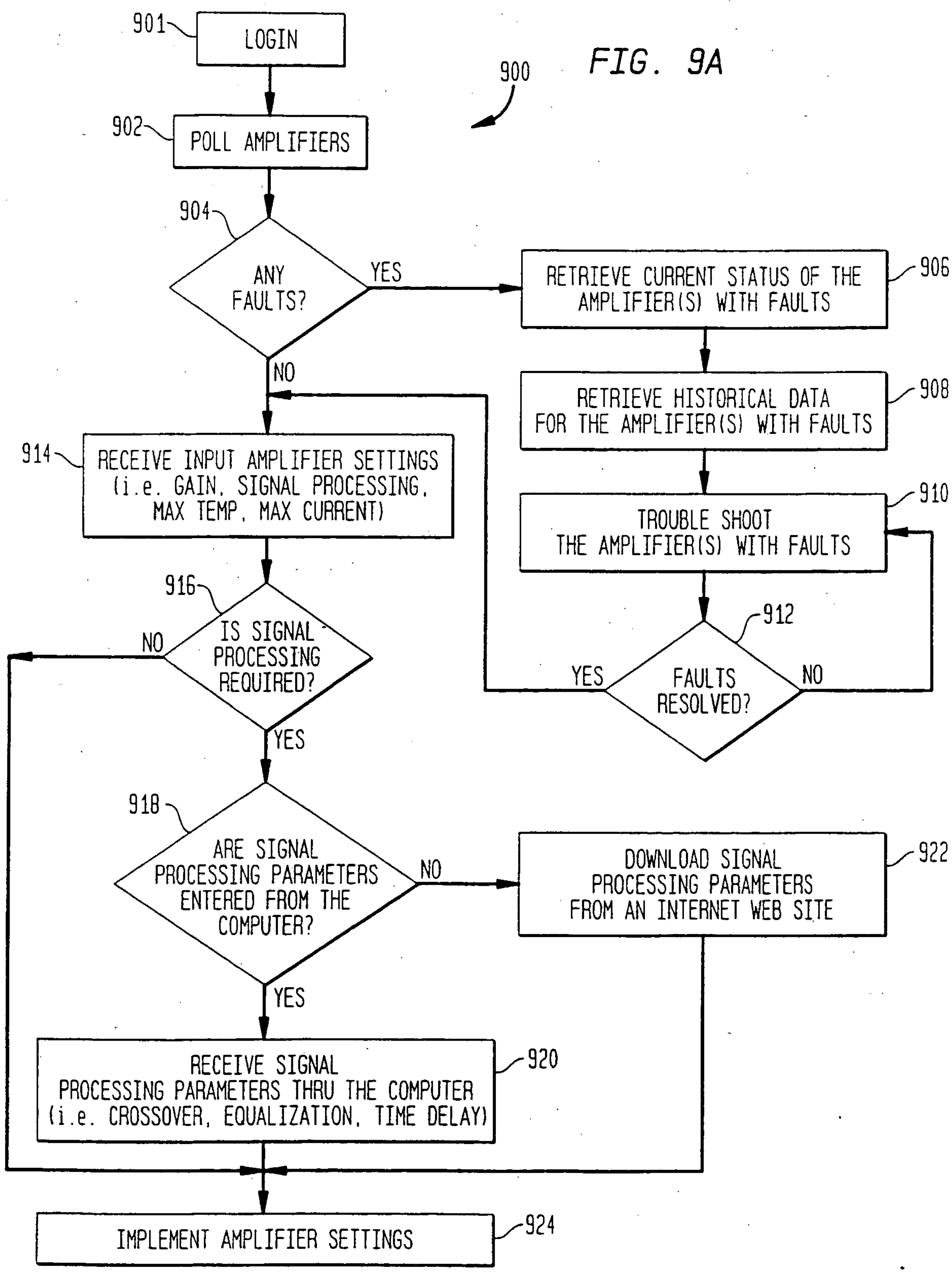

901
LOGIN
900
902
POLL AMPLIFIERS
904
ANY FAULTS?
YES
NO
906
RETRIEVE CURRENT STATUS OF THE AMPLIFIER(S) WITH FAULTS
908
RETRIEVE HISTORICAL DATA FOR THE AMPLIFIER(S) WITH FAULTS
910
TROUBLE SHOOT THE AMPLIFIER(S) WITH FAULTS
912
FAULTS RESOLVED?
YES
NO
914
RECEIVE INPUT AMPLIFIER SETTINGS (i.e. GAIN, SIGNAL PROCESSING, MAX TEMP, MAX CURRENT)
916
IS SIGNAL PROCESSING REQUIRED?
NO
YES
918
ARE SIGNAL PROCESSING PARAMETERS ENTERED FROM THE COMPUTER?
NO
YES
922
DOWNLOAD SIGNAL PROCESSING PARAMETERS FROM AN INTERNET WEB SITE
920
RECEIVE SIGNAL PROCESSING PARAMETERS THRU THE COMPUTER (i.e. CROSSOVER, EQUALIZATION, TIME DELAY)
924
IMPLEMENT AMPLIFIER SETTINGS

FIG. 9B

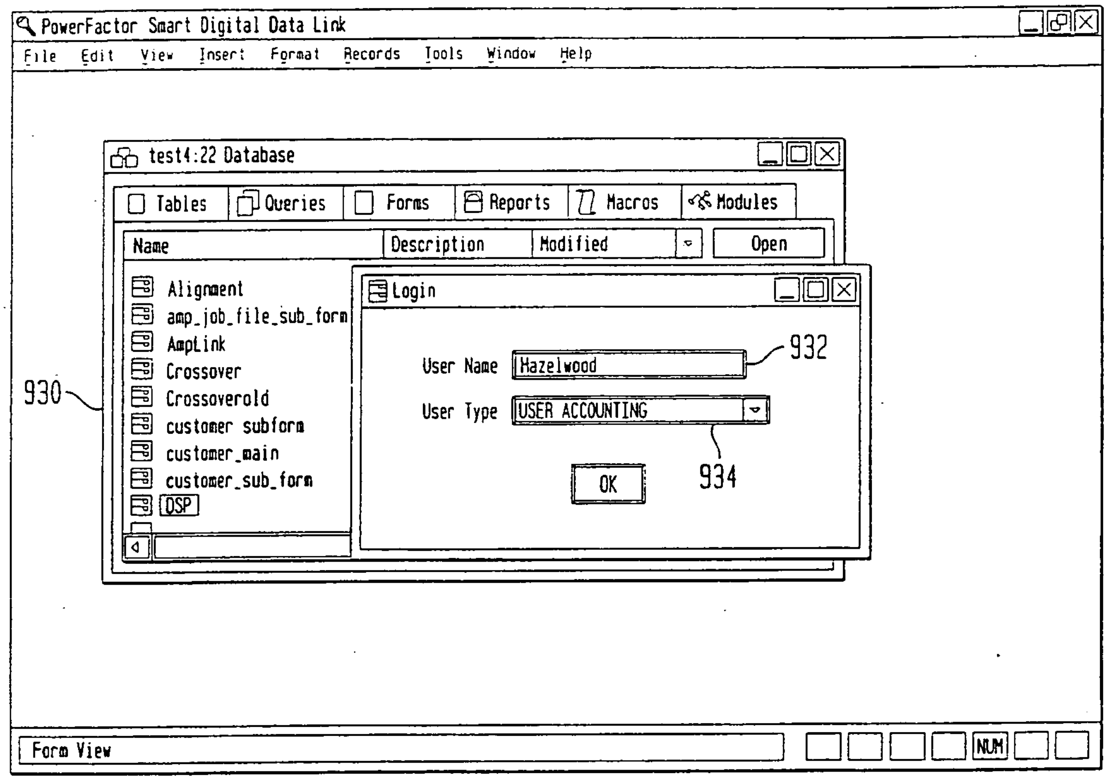

PowerFactor Smart Digital Data Link
File Edit View Insert Format Records Tools Window Help
test4:22 Database
Tables Queries Forms Reports Macros Modules
Name Description Modified Open
Alignment
amp_job_file_sub_form
AmpLink
Crossover
Crossoverold
customer subform
customer_main
customer_sub_form
DSP
Login
User Name Hazelwood
User Type USER ACCOUNTING
OK
930
932
934
Form View
NUM

FIG. 9C

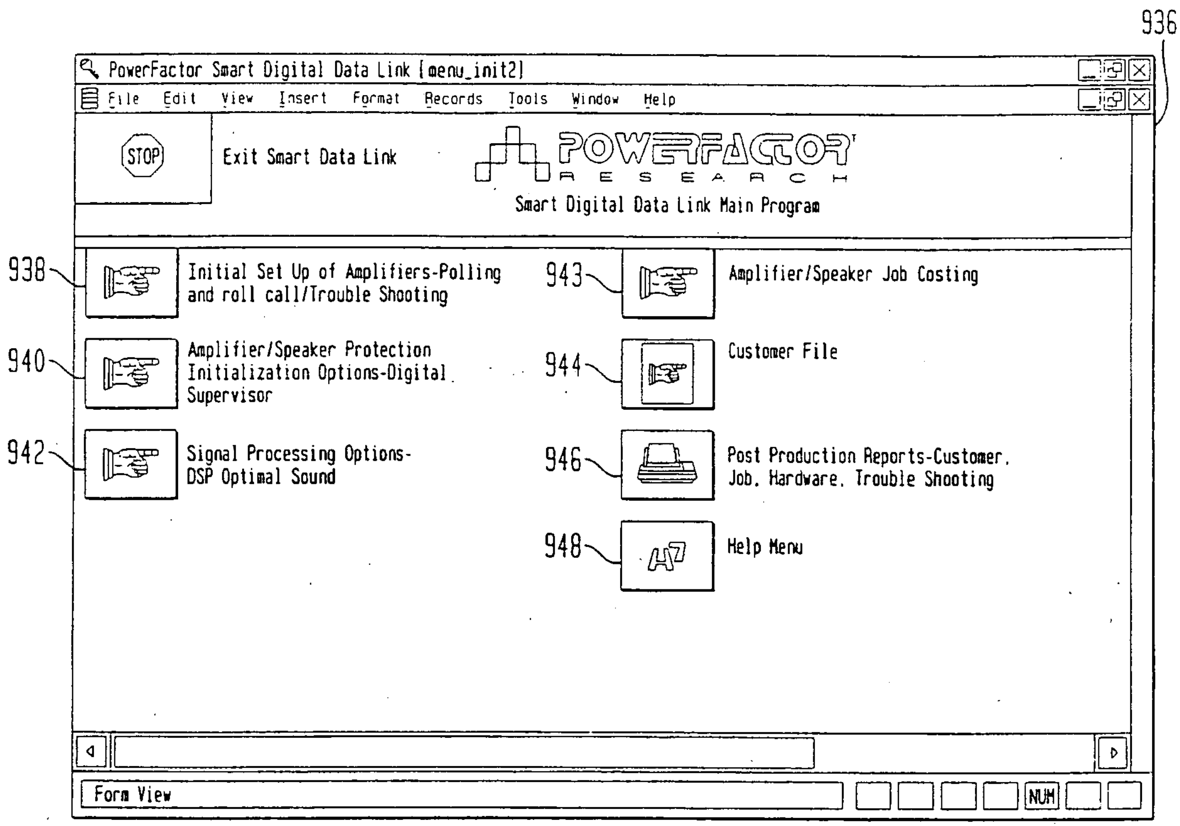

936
PowerFactor Smart Digital Data Link [menu_init2]
File Edit View Insert Format Records Tools Window Help
STOP
Exit Smart Data Link
POWERFACTOR RESEARCH
Smart Digital Data Link Main Program
938
Initial Set Up of Amplifiers-Polling and roll call/Trouble Shooting
940
Amplifier/Speaker Protection Initialization Options-Digital Supervisor
942
Signal Processing Options-DSP Optimal Sound
943
Amplifier/Speaker Job Costing
944
Customer File
946
Post Production Reports-Customer, Job, Hardware, Trouble Shooting
948
Help Menu
Form View
NUM

FIG. 9D

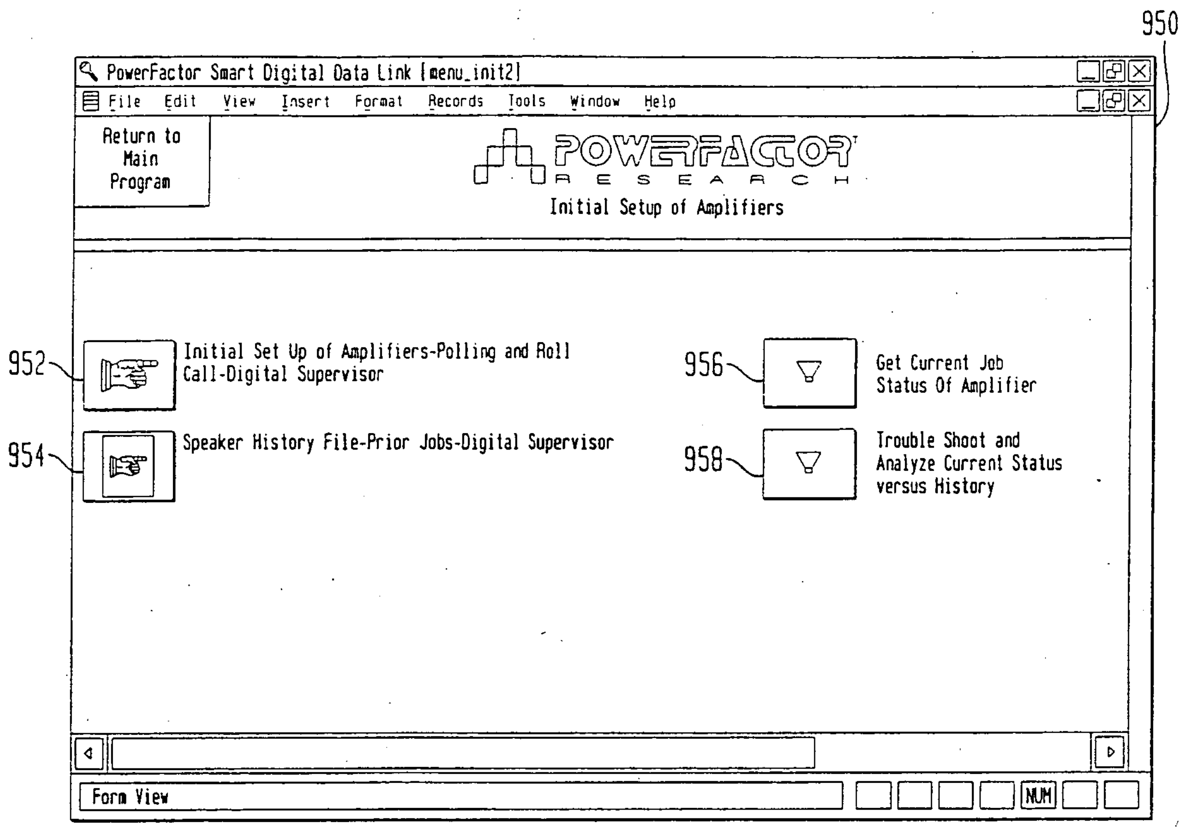

950
PowerFactor Smart Digital Data Link [menu_init2]
File Edit View Insert Format Records Tools Window Help
Return to Main Program
POWERFACTOR
RESEARCH
Initial Setup of Amplifiers
952
Initial Set Up of Amplifiers-Polling and Roll Call-Digital Supervisor
954
Speaker History File-Prior Jobs-Digital Supervisor
956
Get Current Job Status Of Amplifier
958
Trouble Shoot and Analyze Current Status versus History
Form View
NUM

FIG. 9E

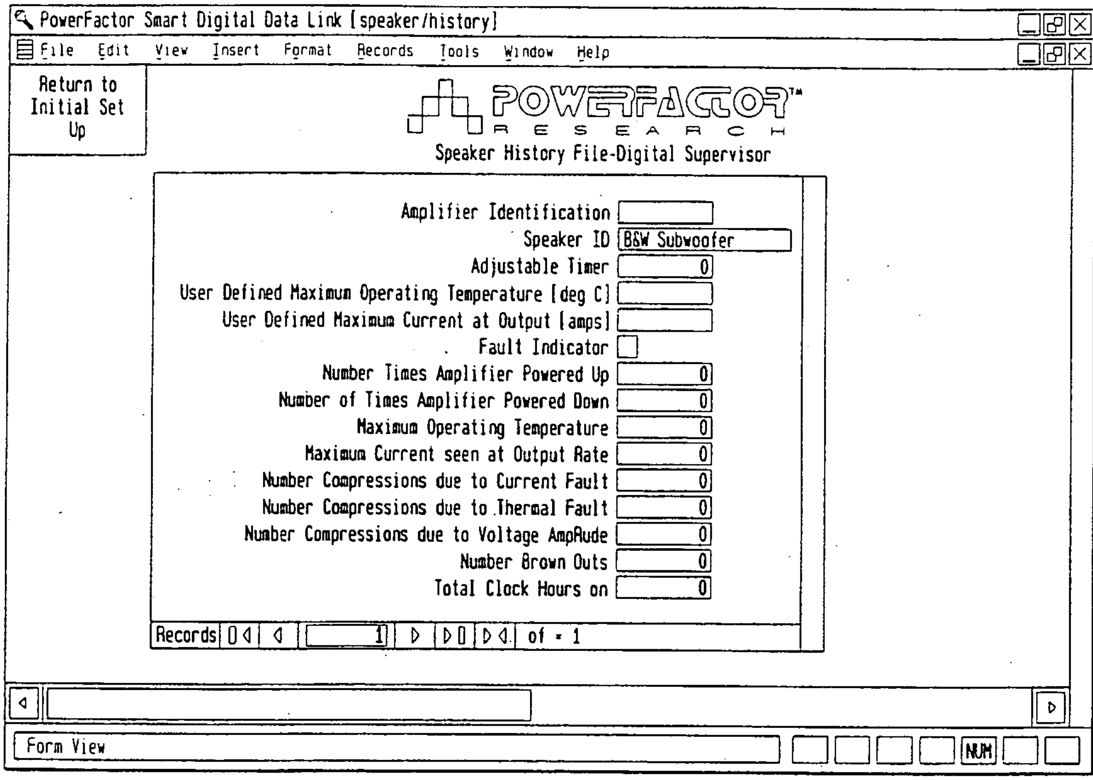
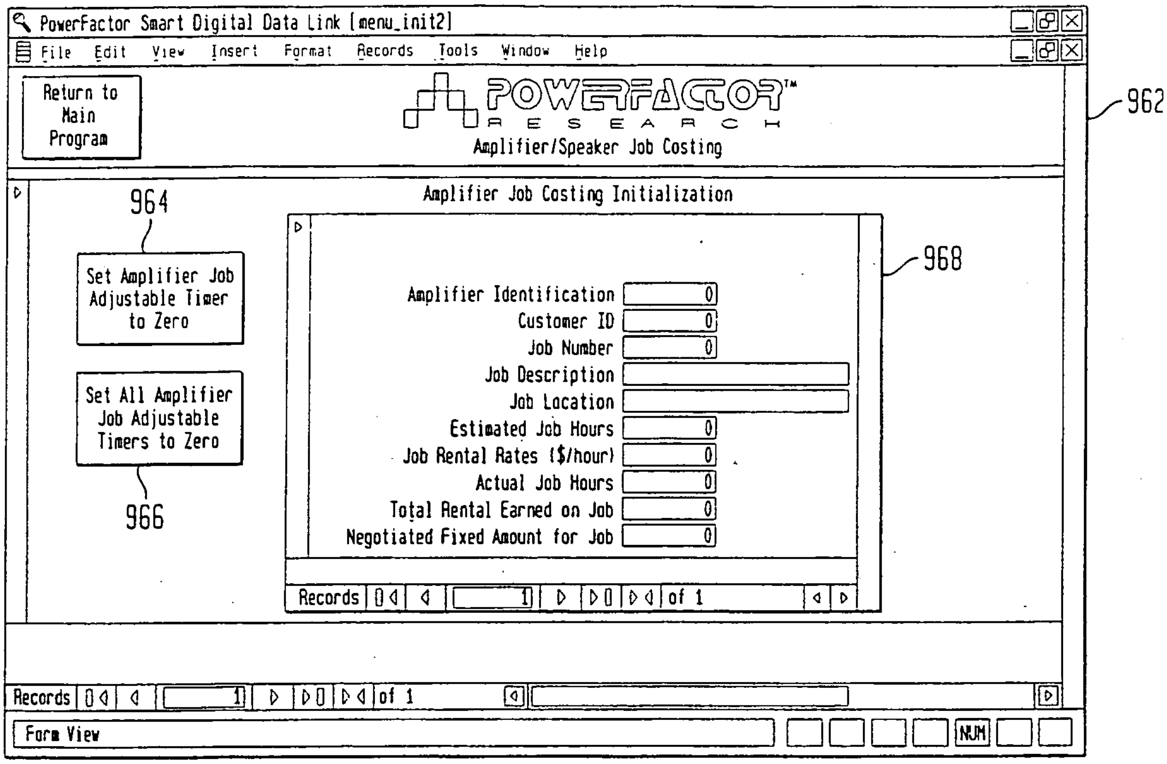

PowerFactor Smart Digital Data Link [speaker/history]
File Edit View Insert Format Records Tools Window Help
Return to Initial Set Up
POWERFACTOR™ RESEARCH
Speaker History File-Digital Supervisor
Amplifier Identification
Speaker ID
B&W Subwoofer
Adjustable Timer
0
User Defined Maximum Operating Temperature [deg C]
User Defined Maximum Current at Output [amps]
Fault Indicator
Number Times Amplifier Powered Up
0
Number of Times Amplifier Powered Down
0
Maximum Operating Temperature
0
Maximum Current seen at Output Rate
0
Number Compressions due to Current Fault
0
Number Compressions due to Thermal Fault
0
Number Compressions due to Voltage AmpRude
0
Number Brown Outs
0
Total Clock Hours on
0
Records
1
of 1
Form View
NUM
960

FIG. 9F



SUBSTITUTE SHEET (RULE 26)

FIG. 10

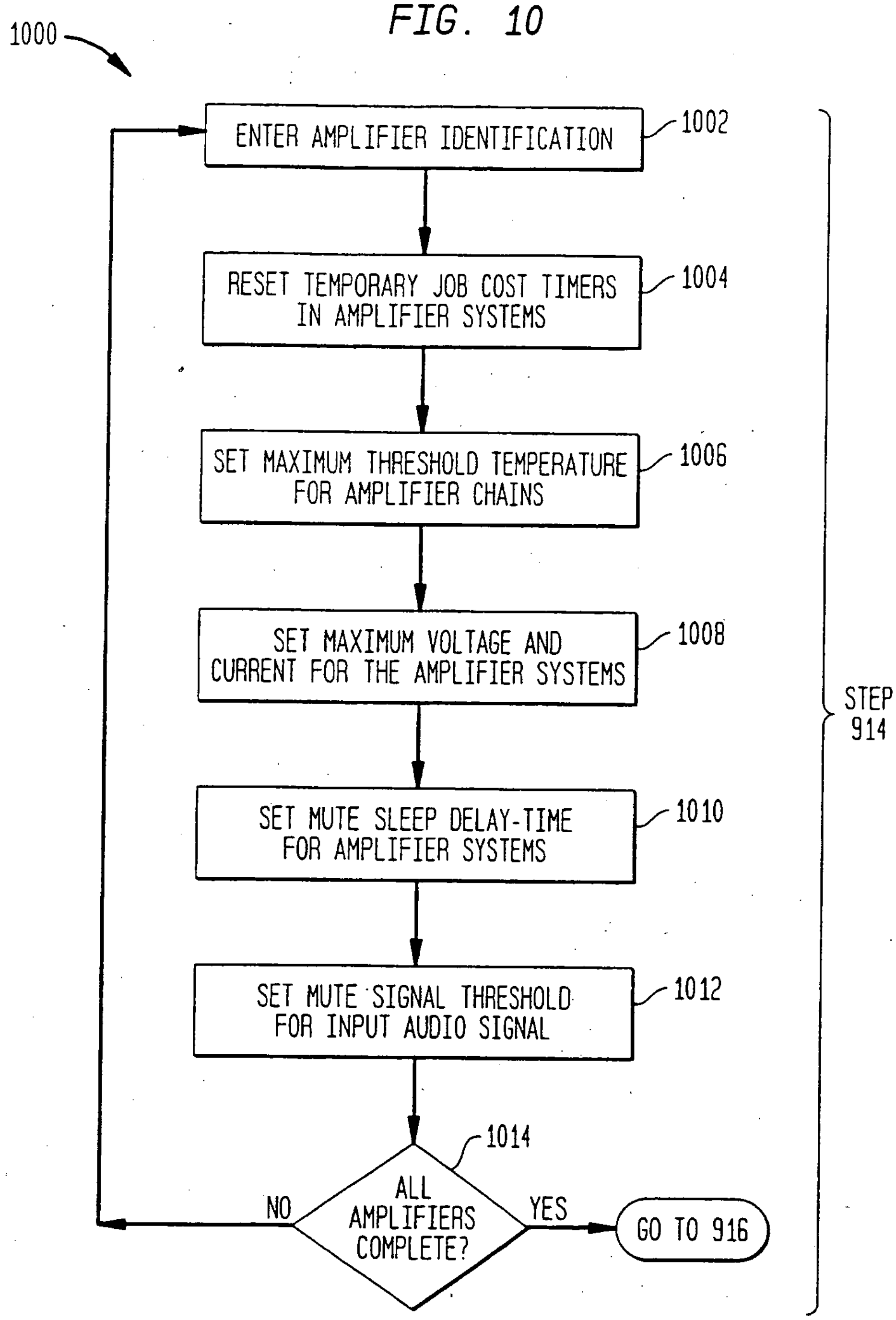

1000
ENTER AMPLIFIER IDENTIFICATION
1002
RESET TEMPORARY JOB COST TIMERS IN AMPLIFIER SYSTEMS
1004
SET MAXIMUM THRESHOLD TEMPERATURE FOR AMPLIFIER CHAINS
1006
SET MAXIMUM VOLTAGE AND CURRENT FOR THE AMPLIFIER SYSTEMS
1008
SET MUTE SLEEP DELAY-TIME FOR AMPLIFIER SYSTEMS
1010
SET MUTE SIGNAL THRESHOLD FOR INPUT AUDIO SIGNAL
1012
1014
ALL AMPLIFIERS COMPLETE?
NO
YES
GO TO 916
STEP 914

FIG. 11A

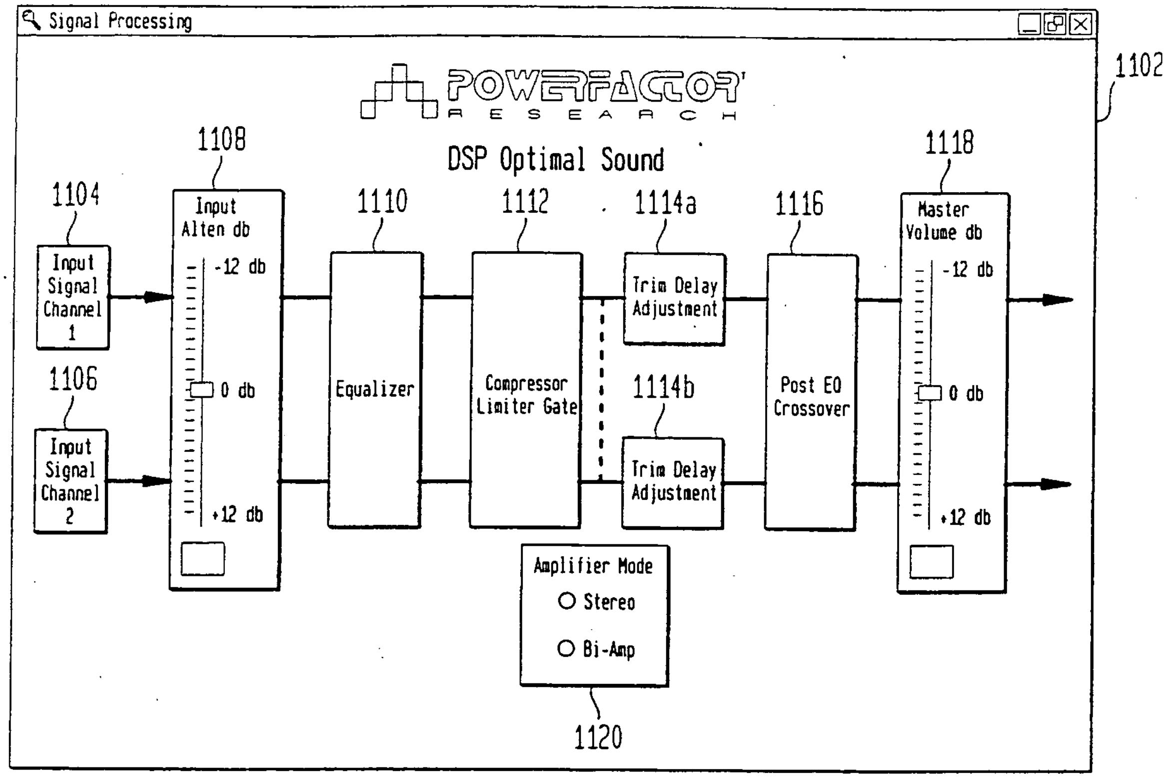

Signal Processing
POWERFACTOR RESEARCH
DSP Optimal Sound
1102
1104
Input Signal Channel 1
1106
Input Signal Channel 2
1108
Input Alten db
-12 db
0 db
+12 db
1110
Equalizer
1112
Compressor Limiter Gate
1114a
Trim Delay Adjustment
1114b
Trim Delay Adjustment
1116
Post EQ Crossover
1118
Master Volume db
-12 db
0 db
+12 db
Amplifier Mode
Stereo
Bi-Amp
1120

FIG. 11B

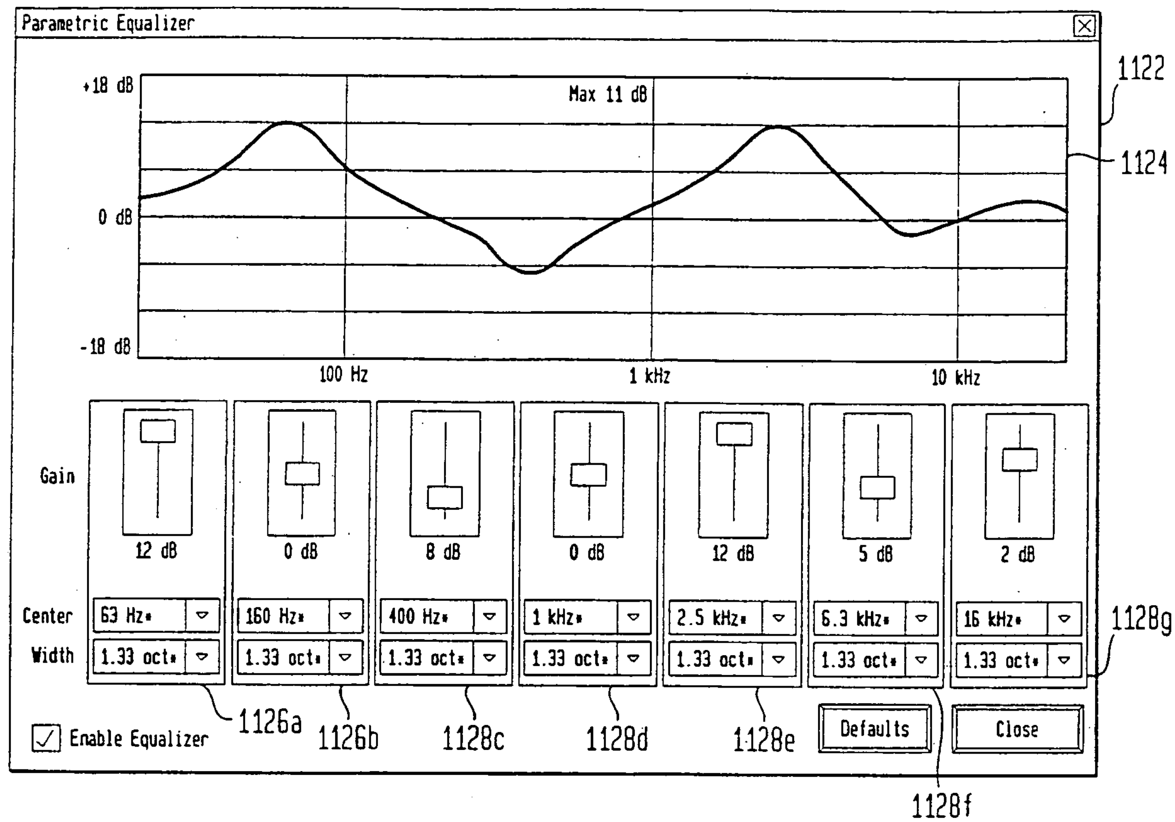

Parametric Equalizer
+18 dB
0 dB
-18 dB
Max 11 dB
100 Hz
1 kHz
10 kHz
Gain
12 dB
0 dB
8 dB
0 dB
12 dB
5 dB
2 dB
Center
Width
63 Hz
160 Hz
400 Hz
1 kHz
2.5 kHz
6.3 kHz
16 kHz
1.33 oct
Enable Equalizer
Defaults
Close
1122
1124
1126a
1126b
1128c
1128d
1128e
1128f
1128g

FIG. 11C

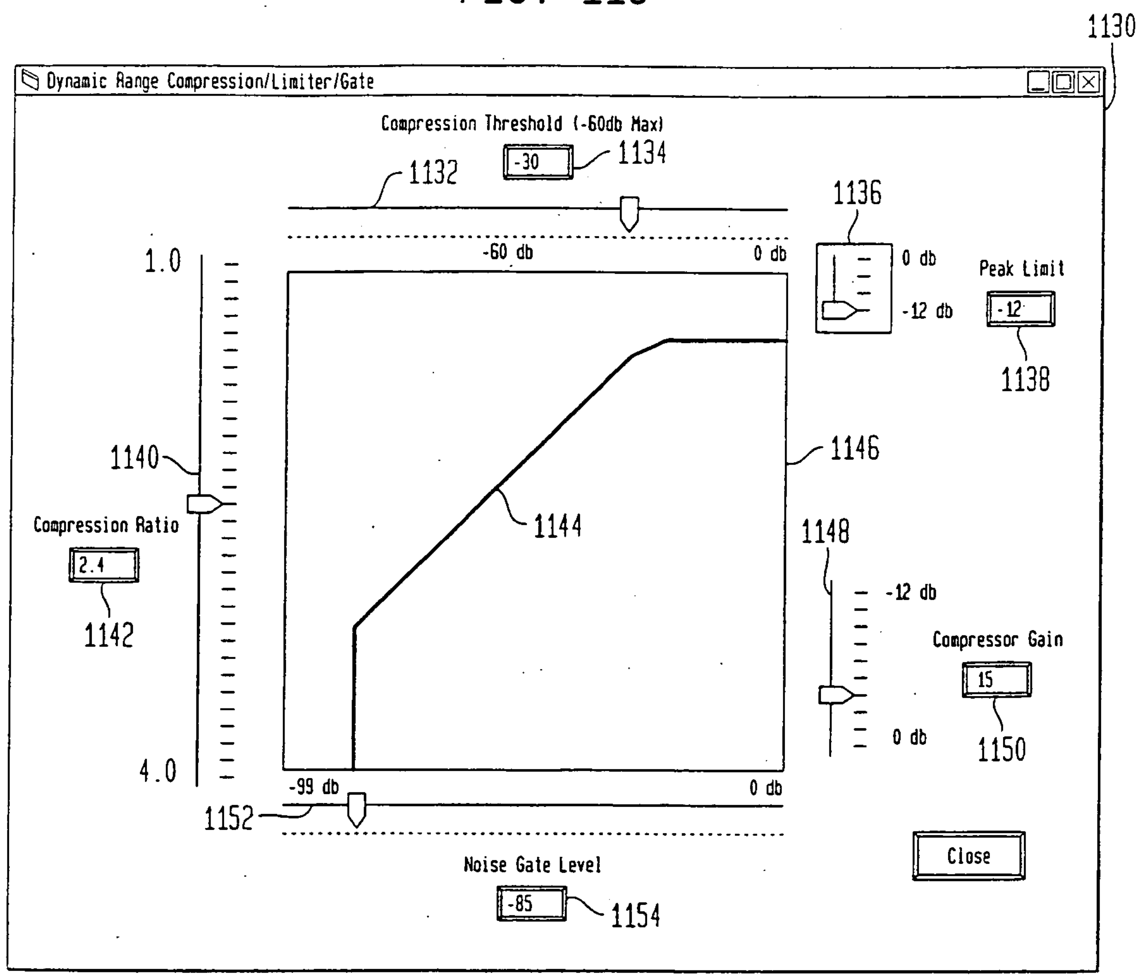

1130
Dynamic Range Compression/Limiter/Gate
Compression Threshold (-60db Max)
-30
1134
1132
1136
-60 db
0 db
0 db
-12 db
Peak Limit
-12
1138
1.0
1140
1146
1144
Compression Ratio
2.4
1142
1148
-12 db
Compressor Gain
15
1150
0 db
4.0
-99 db
0 db
1152
Close
Noise Gate Level
-85
1154

FIG. 11D

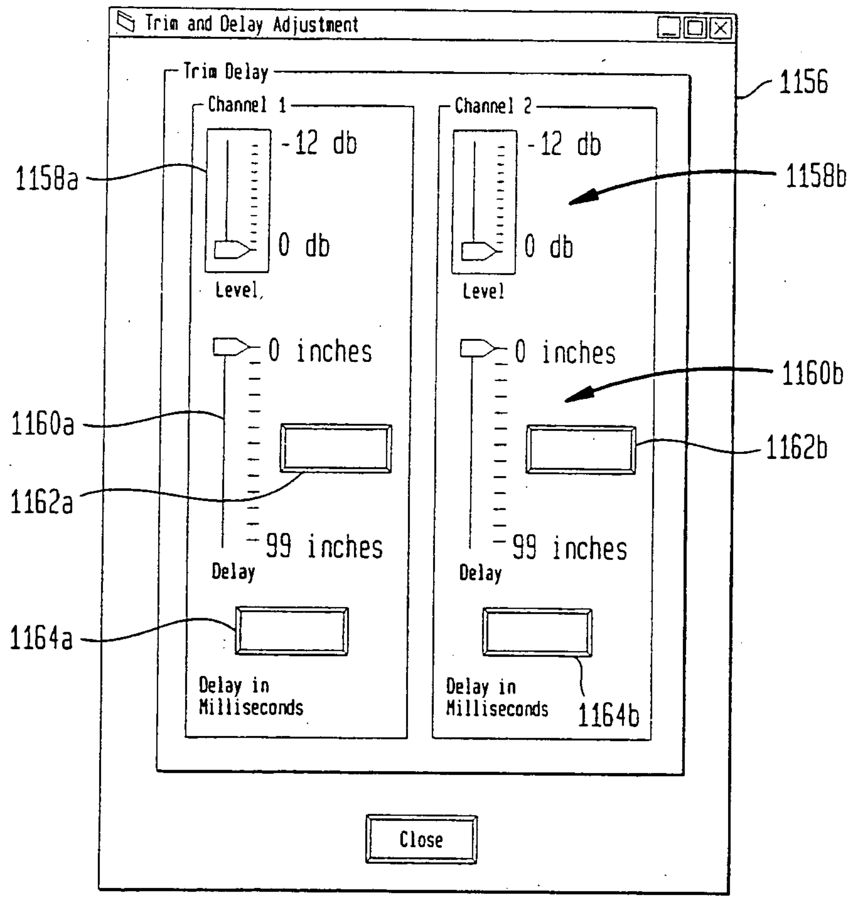

Trim and Delay Adjustment
Trim Delay
Channel 1
Channel 2
-12 db
0 db
Level
0 inches
99 inches
Delay
Delay in Milliseconds
Close
1156
1158a
1158b
1160a
1160b
1162a
1162b
1164a
1164b

FIG. 11E

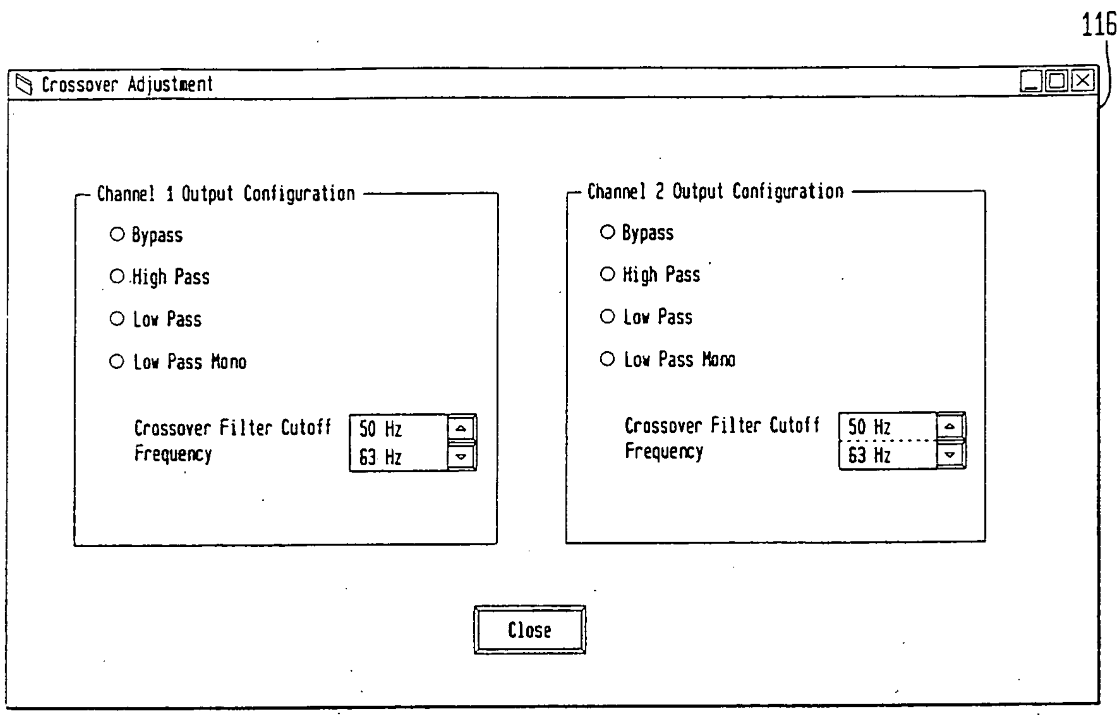

1166
Crossover Adjustment
Channel 1 Output Configuration
Bypass
High Pass
Low Pass
Low Pass Mono
Crossover Filter Cutoff Frequency
50 Hz
63 Hz
Channel 2 Output Configuration
Bypass
High Pass
Low Pass
Low Pass Mono
Crossover Filter Cutoff Frequency
50 Hz
63 Hz
Close

FIG. 12

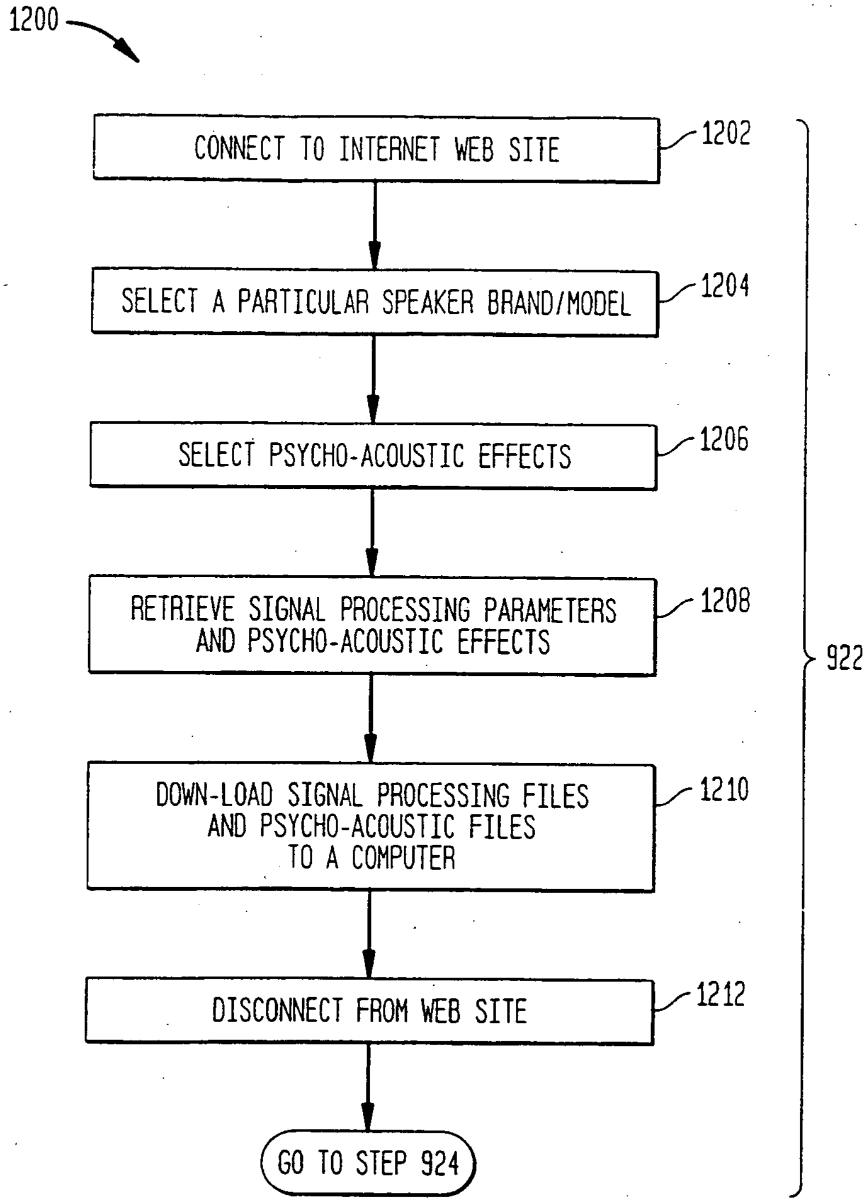
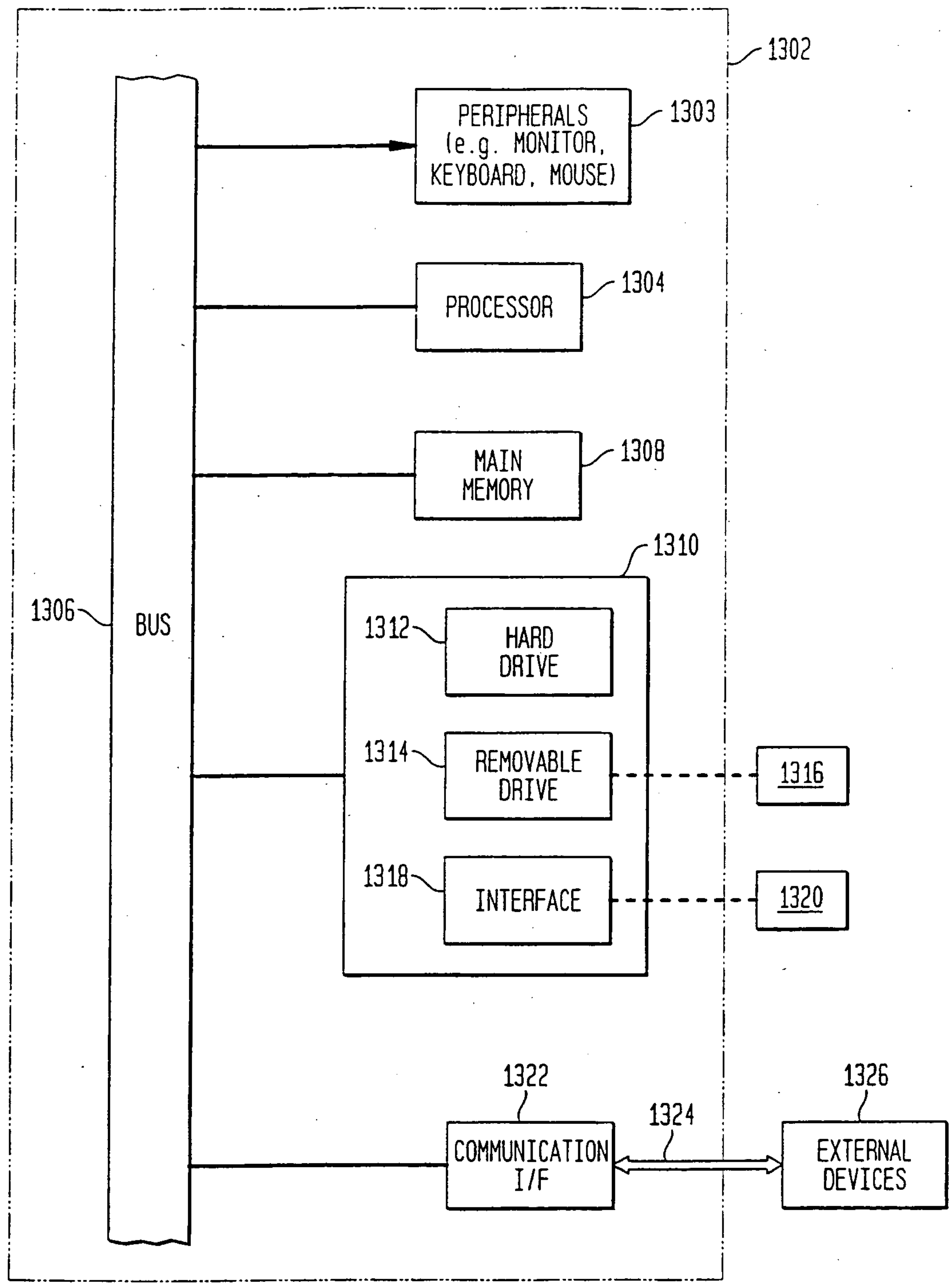

1200
CONNECT TO INTERNET WEB SITE
1202
SELECT A PARTICULAR SPEAKER BRAND/MODEL
1204
SELECT PSYCHO-ACOUSTIC EFFECTS
1206
RETRIEVE SIGNAL PROCESSING PARAMETERS AND PSYCHO-ACOUSTIC EFFECTS
1208
DOWN-LOAD SIGNAL PROCESSING FILES AND PSYCHO-ACOUSTIC FILES TO A COMPUTER
1210
DISCONNECT FROM WEB SITE
1212
GO TO STEP 924
922

FIG. 13


1302
PERIPHERALS (e.g. MONITOR, KEYBOARD, MOUSE)
1303
PROCESSOR
1304
MAIN MEMORY
1308
1310
1306
BUS
1312
HARD DRIVE
1314
REMOVABLE DRIVE
1316
1318
INTERFACE
1320
1322
COMMUNICATION I/F
1324
1326
EXTERNAL DEVICES

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US00/26819

A. CLASSIFICATION OF SUBJECT MATTER

IPC(7) :Please See Extra Sheet.
US CL : 381/55, 58, 59, 96, 120; 84/615; 330/2
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 381/55, 58, 59, 96, 120; 84/615

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 5,255,324 A (BREWER et al) 19 OCTOBER 1993 , COL. 3, LINES 56-58, COL. 7, LINES 7-37, COL. 4, LINES 17-22, COLUMNS 3-4, LINES 60-2	1, 2, 4
Y	Figures 1and 2, col. 3, 56-58 and 7-37, col. 4, lines 17-22, columns 3-4, lines 60-2, column 4, lines 38-48, column 7, lines 12-44	9, 11, 13, 29, 30, 31
Y	US 4,688,002 A (WINGATE) 18 AUGUST 1987, FIGURE 1, REFERENCE 10, COL. 6, LINES 40-46	1, 9, 11, 13
X, E	US 6,160,213 A (ARNOLD et al.) 12 DECEMBER 2000, COL. 2, LINES 45-61, COL. 14, LINES 49-52, COL. 15, LINES 55-67, COL. 16, LINES 1-18, COL. 18, LINES 35-39	90, 112, 113, 114, 118

[X] Further documents are listed in the continuation of Box C. [ ] See patent family annex.

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search 08 JANUARY 2001	Date of mailing of the international search report 05 FEB 2001
Name and mailing address of the ISA/US Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231 Facsimile No. (703) 305-3230	Authorized officer LAURA A. GRIER Karen L. Ward Telephone No. (703) 305-4700

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US00/26819

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X — Y	US 5,652,542 A (FINK) 29 JULY 1997, Abstract-57, col. 5, lines 17-67, col. 5, lines 17-30, column 6, lines 36-52, FIGURES 1 AND 2, REFERENCES 202, 204, 260, 100, AND 300 AND COLUMN 4, LINES 62-67, COL. 5, LINES 36-40, COL. 6, LINES 1-4, COL. 5, LINES 56-67, COL. 1, LINES 24-42, COL. 2, LINES 60-62 AND COL. 3, LINES 16-24,	39, 75, 107 — 1, 2, 3, 8, 10, 11, 29,30,34, 36, 38, 48, 52
Y	US 5,410,265 A (JAIN et al) 25 APRIL 1995, COL. 2, LINES 18-22	52

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US00/26819

A. CLASSIFICATION OF SUBJECT MATTER:
IPC (7):

H03G 11/00
H04R 29/00
H04R 3/00
H03F 21/00
G10H 1/18
G01R 19/00

Form PCT/ISA/210 (extra sheet) (July 1998)★